MediaPack™ User's Manual
for MP-124, MP-108, MP-104 and MP-102
VoIP H.323 Gateways

Version 4.2 Beta

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Abbreviations and Terminology

Each abbreviation, unless widely used, is spelled out in full when first used, and only Industry standard terms are used throughout this manual. The $ symbol indicates hexadecimal notation.

Documentation

The following AudioCodes manuals should be used for additional information:

"MP-Series H.323 Release Notes", Document #: LTRT-00652

"VoPLib Reference Library User's Manual", Document #: LTRT-00744.

Notice

This **MP-1xx/H.323** User's Manual describes the AudioCodes MediaPack Series **MP-124** 24 port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port, referred to collectively as the **MP-1xx**, supported by software version 4.2 Beta.

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FCC Notice to Users

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

Safety Notice

Installation and service of this Gateway must only be performed by authorized, qualified service personnel.

Telecommunication Safety

The safety status of each port on the Gateway is declared and detailed in the table below:

Ports	Safety Status
Ethernet (100 Base-T)	SELV
FXS	TNV-3

TNV-3: Circuit whose normal operating voltages exceeds the limits for an SELV circuit under normal operating conditions and on which over voltages from Telecommunication Networks are possible

SELV: Safety extra low voltage circuit.

Declaration of Conformity

We AudioCodes Ltd

Declare under our sole responsibility that the products:

MP-1xx/FXS

To which this declaration relates, is in conformity with the following standards: EN 55022 1998, EN 50024 1998, EN 60950 1992 + Amendments 1, 2, 3 & 4

As described in the European Directives: 89/336 (EMC),
73/23 (Safety),
93/68 (Safety).

Yehud, Israel, 28 Jun 2001
I. Zusmanovich
Compliance Eng.

Este producto està en conformidad con la directiva Europea 89/336, 73/23, 93/68
Dette produkt er i overensstemmelse med Europæiske Direktiver 89/336, 73/23, 93/68
Dieses Produkt ist konform mit der europäischen Richtlinie 89/336, 73/23, 93/68
Ce produit est conforme aux exigences de la Directive européenne 89/336, 73/23, 93/68
Questo prodotto è conforme con la normativa europea 89/336, 73/23, 93/68
Este producto está em conformidade com as Diretrizes Européia 89/336, 73/23, 93/68
Tuote on eurooppalaisen säännöstön mukainen 89/336, 73/23, 93/68
Denna product följer europeiska direktiv 89/336, 73/23, 93/68
Το προιόν, είναι σύμφωνο με τους κανονισμούς της Ευρωπαικής Κοινώτητας 89/336, 73/23, 93/68
Tæki þetta er í samræmi við tilskipun Evrópusambandsins 89/336, 73/23, 93/68
Dette produktet er i samhørighet med det Europeiske Direktiv 89/336, 73/23, 93/68

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS Gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO Gateways.

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Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS Gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO Gateways.

User's Manual for
MP-102, MP-104, MP-108 and MP-124
H.323 Media Gateways

1 MP-1xx/H.323 Overview

1.1 Introduction

This document provides the User with the information about installation, configuration and operation of the **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port VoIP Media Gateways. As these units have similar functionality, except for the number of channels and some minor features, they are referred to collectively as the **MP-1xx**. It is expected that the readers are familiar with regular telephony and data networking concepts.

1.2 Gateway Description

The **MP-1xx** telephony Media Gateway provides excellent voice quality and optimized packet voice streaming over IP networks. The product enables voice, fax and data traffic to be sent over the same IP network. It is based on AudioCodes award-winning, field-proven TrunkPack design using the AudioCodes well-established DSP voice compression technology.

The **MP-1xx** incorporates up to 24 analog ports for connection, either directly to an enterprise PBX (**MP-10x**/FXO), to phones, or to fax (**MP-1xx**/FXS), supporting up to 24 simultaneous VoIP calls.

Additionally, the **MP-1xx** units are equipped with a 10/100 Base-T Ethernet port for connection to the LAN.

The **MP-1xx** Gateways are best suited for small to medium size enterprises, branch offices or for residential Media Gateway solutions.

The **MP-1xx** Gateways enable Users to make free local or international telephone/fax calls between the distributed company offices, using their existing telephones/fax. These calls are routed over the existing IP Internet or Intranet corporate data networks ensuring that voice traffic takes the minimum of space on the data network.

The **MP-1xx** Gateways are very compact devices, designed to be installed either as a desk-top unit (refer to

Figure 1-2) or installed in a 19-inch rack (refer to Figure 2-1).

The **MP-1xx** supports H.323 ITU, SIP, MEGACO (H.248) and MGCP protocols, enabling the deployment of "voice over packet" solutions in environments where each enterprise or residential location is provided with a simple Media Gateway.

This provides the enterprise with a telephone connection (e.g., RJ-11), and the ability to transmit the voice and telephony signals over a packet network.

Additionally, for emergency use, the **MP-10x/FXS Gateway** provides a Life Line, connected to the unused pins on port #4 (or port #2 for **MP-102**), with a relay to an analog line, even if the **Gateway** is powered off.

The layout diagram, Figure 1-5 on page 15, illustrates a typical **MP-108** and **MP-104** or **MP-102** VoIP application.

Figure 1-1: MP-124 VoIP Gateway

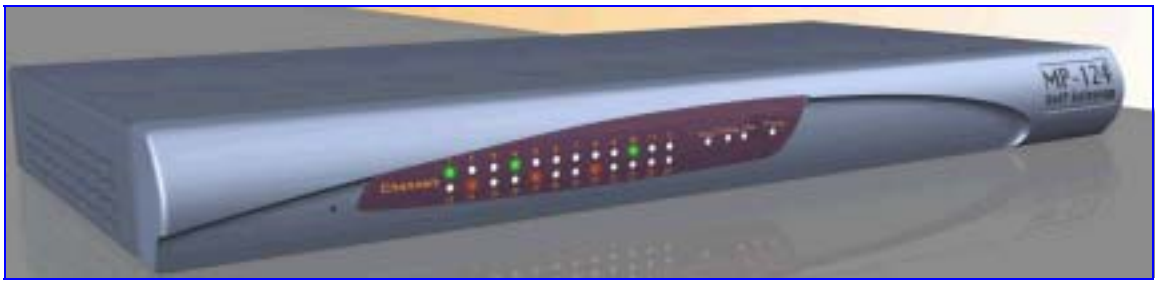


Figure 1-2: MP-108 Front View

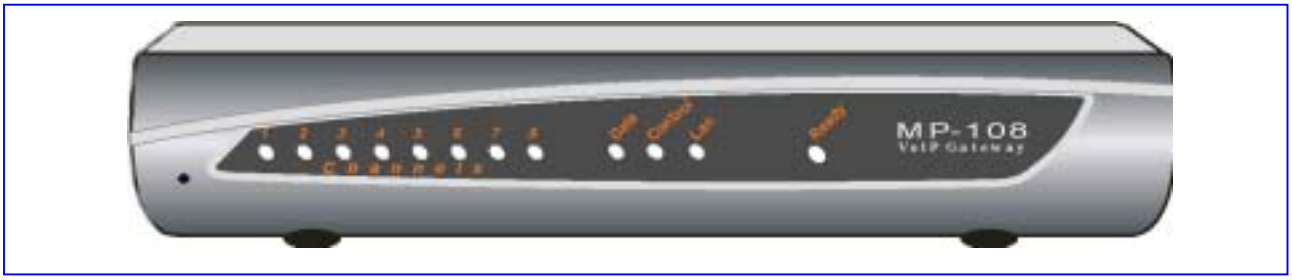


Figure 1-3: MP-104 Front View

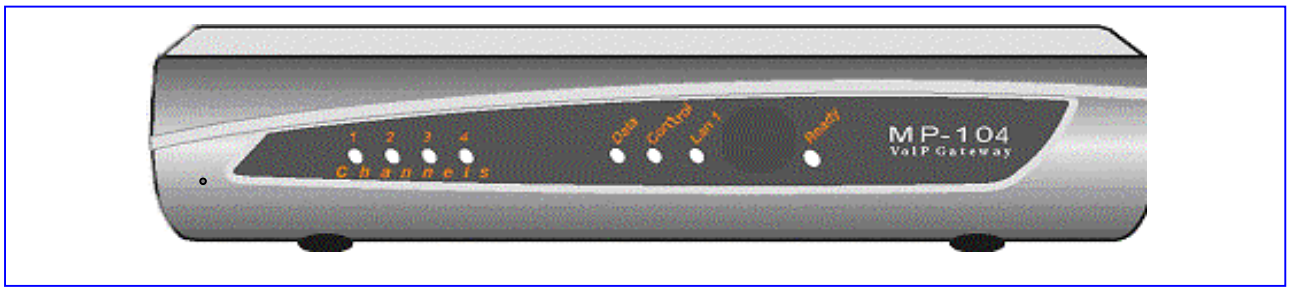


Figure 1-4: MP-102 Front View

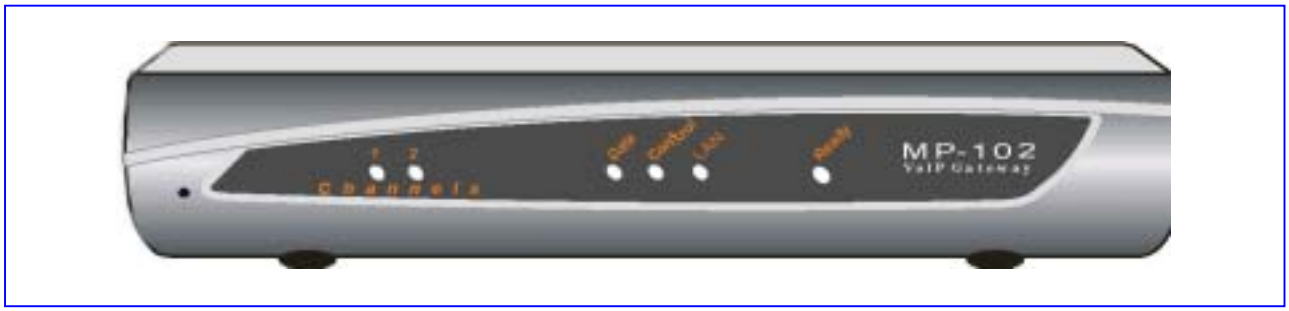
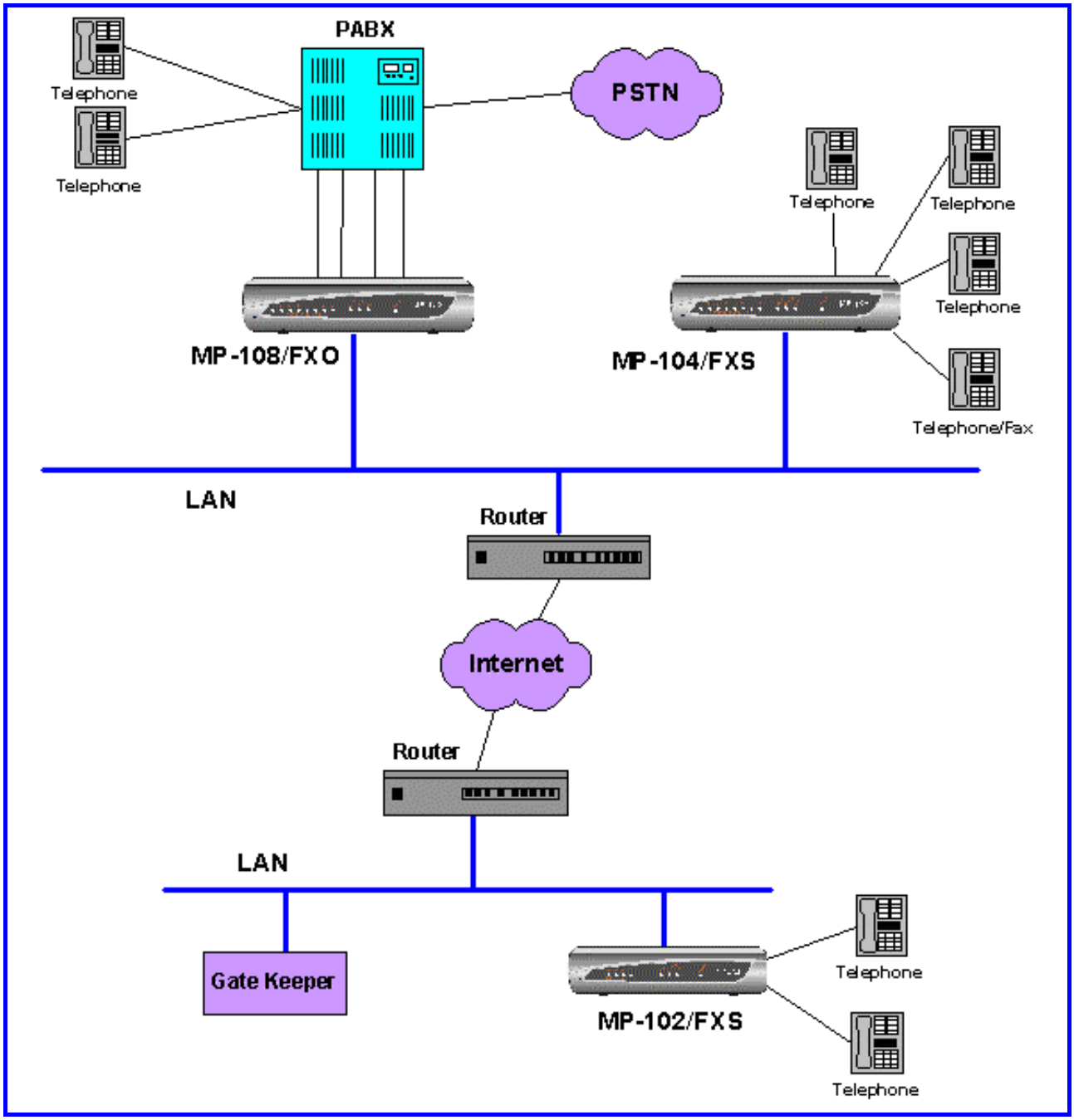


Figure 1-5: Typical MP-1xx VoIP Application



1.3 MP-1xx Key Features

- High quality Voice, Data and Fax over IP networks.
- **MP-124** supports up to 24 analog telephone loop start FXS ports as shown in Figure 1-1 on page 14.
- **MP-108** supports up to 8 analog telephone loop start FXS or FXO ports as shown in
- Figure 1-2 on page 14.
- **MP-104** supports up to 4 analog telephone loop start FXS or FXO ports as shown in
- Figure 1-3 on page 14.
- **MP-102** supports up to 2 analog telephone loop start FXS ports as shown in
- Figure 1-4 on page 14.
- Connected to the IP network via a 10/100 Base-T Ethernet interface.
- Coders include: G.711, G.723.1, G.726, G.727, G.729A and NetCoder at 6.4 to 8.8 kbps, selectable per channel.
- T.38 Fax with superior performance (round trip delay up to 9 sec).
- Compliant with H.323 (Version 4), SIP, MEGACO (H.248) and MGCP.
- Life Line, connected to the unused pins on port #4 (or port #2 for **MP-102**), with a relay to an analog line, even if the **MP-10x/FXS** is powered off.
- LEDs on the front and rear panels provide information on the operating Media Gateway status and of the network interface.
- Restart button on the Front panel restarts the **MP-1xx** Gateway
- **MP-10x** compact, rugged enclosure providing up to 8 analog RJ-11 ports within a compact housing of only one-half of a 19-inch rack unit, 1 U high (1.75" or 44.5 mm).
- **MP-124** 19-inch, 1 U rugged enclosure provides up to 24 analog FXS ports, using a single 50 pin Telco connector.
- Mounting option of installing two **MP-10x** Gateways in a single 19-inch rack shelf, one U high (1.75" or 44.5 mm).

2 MP-10x Hardware Installation

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS Gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO Gateways.

2.1 Hardware Installation Procedure

2.1.1 Unpacking

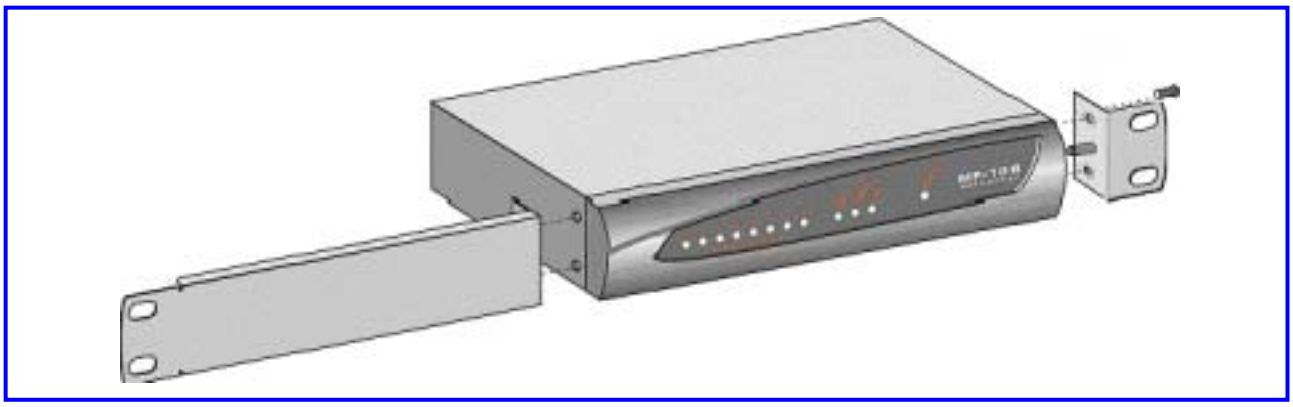
- Open the carton and remove packing materials
- Remove the **MP-10x** Gateway from the carton
- Check that there is no equipment damage
- Check, retain and process any documents
- Notify AudioCodes of any damage or discrepancies
- Retain any diskettes or CDs

Safety Notice

Installation and service of this device must only be performed by authorized, qualified service personnel.

2.1.2 MP-10x Rack Mounting Installation

Figure 2-1: MP-10x Rack Mounting



The **MP-10x** is installed into a standard 19-inch rack by the addition of the 2 brackets supplied, and shown above.

➢ To install the MP-10x, take the following steps:

Step 1 Fasten the short bracket to the right-hand side of the **MP-10x** using the 2 screws provided, as shown in Figure 2-1, and carefully positioning the peg into a convenient ventilation hole in the side of the **MP-10x** box.

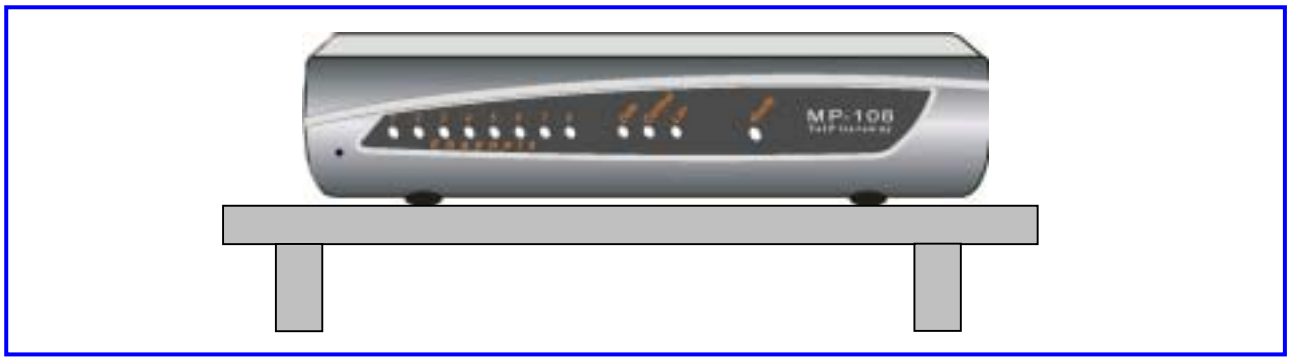
Step 2 Fasten the long bracket to the left-hand side of the **MP-10x** using the 2 screws provided as shown in Figure 2-1, and carefully positioning the peg into a convenient ventilation hole in the side of the **MP-10x** box.

Step 3 Insert the **MP-10x** into the 19-inch rack and fasten the left-hand and right-hand brackets to the vertical tracks of the 19-inch rack, using standard 19-inch rack bolts (not provided).

➢ To connect the cables go to Step 4 (Section 2.1.4 on page 19)

2.1.3 MP-10x Desktop Mounting Installation

Figure 2-2: MP-10x Desktop or Shelf



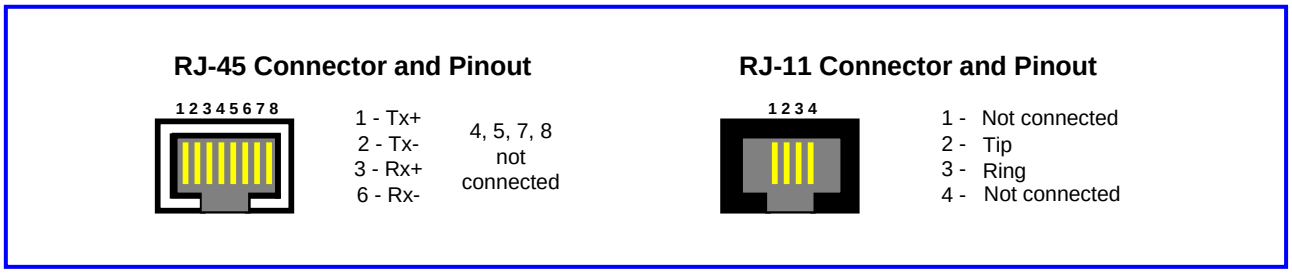
The **MP-10x** is installed on a desk or shelf without additional brackets as shown above.

➢ To connect the cables go to Step 4 (Section 2.1.4 on page 19)

2.1.4 Cable Connections

The RJ-45 (Ethernet) and RJ-11 (Ports) pinouts and connectors are shown in Figure 2-3, and pins are numbered from the left with the latching finger position at the bottom.

Figure 2-3: RJ-45 LAN and RJ-11 Port Connectors and Pinouts


RJ-45 Connector and Pinout
1 2 3 4 5 6 7 8
1 - Tx+
2 - Tx-
3 - Rx+
6 - Rx-
4, 5, 7, 8 not connected
RJ-11 Connector and Pinout
1 2 3 4
1 - Not connected
2 - Tip
3 - Ring
4 - Not connected

➢ **To connect the cables for Desk-top or Rack-mount go to Step 4 below:**

Step 4 When using **MP-10x/FXS** Gateway, insert each of the RJ-11 connectors on the 2-wire line cords of the POTS phones into the RJ-11 sockets on the rear of the Gateway.

When using **MP-10x/FXO** Gateway, insert each of the RJ-11 connectors on the 2-wire line cords coming from PSTN/PBX into the RJ-11 sockets on the rear of the Gateway

Telephone lines and extensions of up to 7,300 m (24,000 ft) can be achieved using regular 24 AWG line cord.

Step 5 Insert the RJ-45 connector on the 10/100 Base-T cable from your LAN to the ETH RJ-45 socket (on the rear of the **MP-10x**) to provide the link to your LAN.

Step 6 Connect the **MP-10x** Gateway to the correct AC power supply, and the installation is now complete.

2.1.5 Installation of the MP-10x/FXS Life Line

Note: The **MP-124 and MP-10x/FXO** Media Gateways do **NOT** support the Life Line.

The **MP-108/FXS and MP-104/FXS** Gateways provide a Life Line connection on port **#4**.

The **MP-102/FXS** Gateway provides a Life Line connection on port **#2**.

This feature provides a wired phone connection to any PSTN or PBX FXS port, upon power-down conditions. When the power outage occurs, the phone that is connected to the Life Line port (see above), on pins #2 and #3, is wired to the PSTN or PBX FXS wires on pins #1 and #4 on the same connector. Therefore, the User of the **MP-10x/FXS**, can use the phone even when the **MP-10x/FXS** is not powered-on. To use this function, the User must utilize a splitter that connects pins #1 and #4 to another source of an FXS port, and pins #2 and #3 to the POTS phone.

The pinout of the Life Line RJ-11 phone connector is as follows:

1 = Life Line TIP

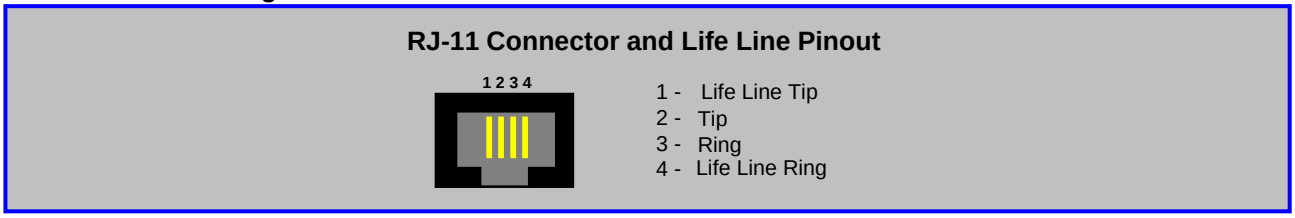
2 = TIP

3 = RING

4 = Life Line RING

Refer to Figure 2-4 below for the RJ-11 connector pinout.

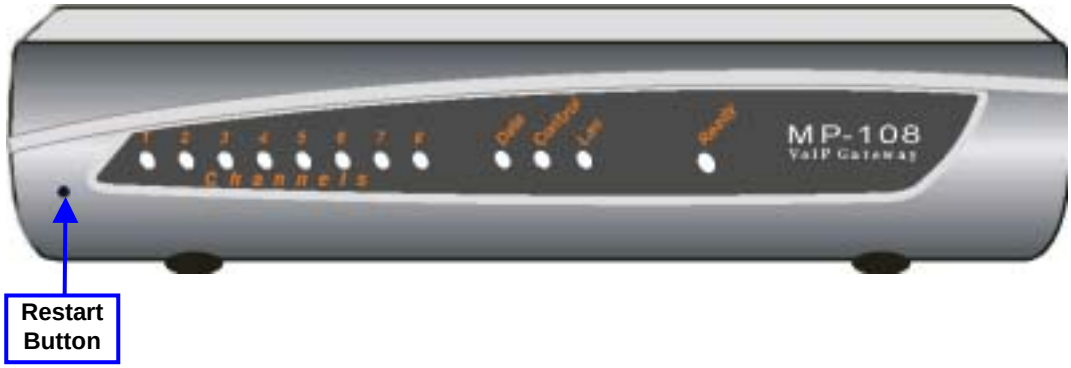
Figure 2-4: RJ-11 Connector and Life Line Pinout for MP-10x/FXS



2.2 Front Panel LED Indicators

The **MP-10x** front panel LEDs indicate the Ethernet LAN status, Data (RTP) activity and state of the **MP-10x** ports.

Figure 2-5: MP-10x Front Panel LED Indicators



Functionality of the Front Panel LEDs for **MP-10x** is explained in Table 2-1.

Table 2-1: Front Panel Network LED Indicators

Label	Type	Color	State	Meaning
LAN	Ethernet Link Status	Green	ON	Valid Connection to 10/100 Base-T hub/switch
		Red	ON	Malfunction
Data	Packet Status	Green	Blinking	Transmitting RTP Packets
		Red	Blinking	Receiving RTP Packets
		Blank	-	No traffic
Control	Control Link	Green	Blinking	Sending and receiving H.323 messages.
		Red		Not supported in current release
Ready	Device Status	Green	ON	Device Powered, Self test OK
		Orange	Blinking	Software Loading/Initialization
		Red	ON	Malfunction

Table 2-2: MP-10x Channel LEDs

MP-10x with 1 to 8 Channels				
Label	**Type**	**Color**	**State**	**Meaning**
Activity	**FXS** Tel Port	Green	ON	Off-Hook/Ringing for Phone Port
	FXO Line Port	Green	ON	Line-Seize/Ringing State for Line Port

2.3 Rear Panel LED Indicators and Connectors

Figure 2-6: Rear Panel LED Indicators and Connectors

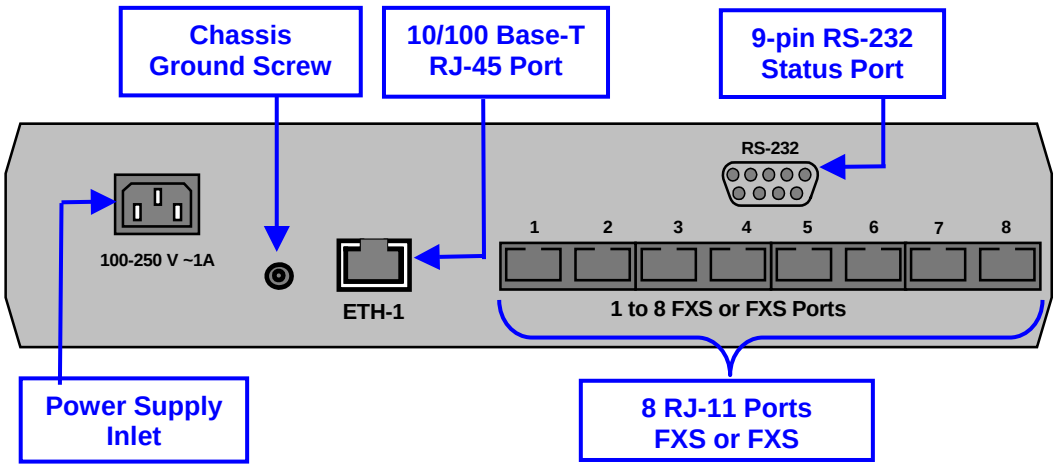

Chassis Ground Screw
10/100 Base-T RJ-45 Port
9-pin RS-232 Status Port
RS-232
100-250 V ~1A
ETH-1
1 2 3 4 5 6 7 8
1 to 8 FXS or FXS Ports
Power Supply Inlet
8 RJ-11 Ports FXS or FXS

Table 2-3: Meaning of Rear Panel LED Indicators

Label	Type	Color	State	Meaning
ETH-1	Ethernet Status	Yellow	ON	Ethernet Port Receiving Data
		Red	ON	Collision

Table 2-4: Explanation of Rear Panel Connectors/Switches

Label	Type	Function	Comment
100-240V ~ 1A	3-pin power inlet	AC input	Connection to external power supply
1 to 8	RJ-11	8 FXS or FXO Ports	**MP-108** 2-wire Loop Start interface
1 to 4	RJ-11	4 FXS or FXO Ports	**MP-104** 2-wire Loop Start interface
1 to 2	RJ-11	2 FXS Ports	**MP-102** 2-wire Loop Start interface
ETH 1	RJ-45	10/100 Base-T Port	Shielded
RS-232	DB-9, DCE	Status Messages	Gateway connects to PC's RS-232 port with a straight cable (refer to Figure 8-1 on page 100).
⏚	Grounding screw	Chassis Ground	**MUST** be securely connected.

Reader's Notes

3 MP-124 Hardware Installation

Safety Notice

Installation and service of this device must only be performed by authorized, qualified service personnel.

3.1 Hardware Installation Procedure

3.1.1 Unpacking

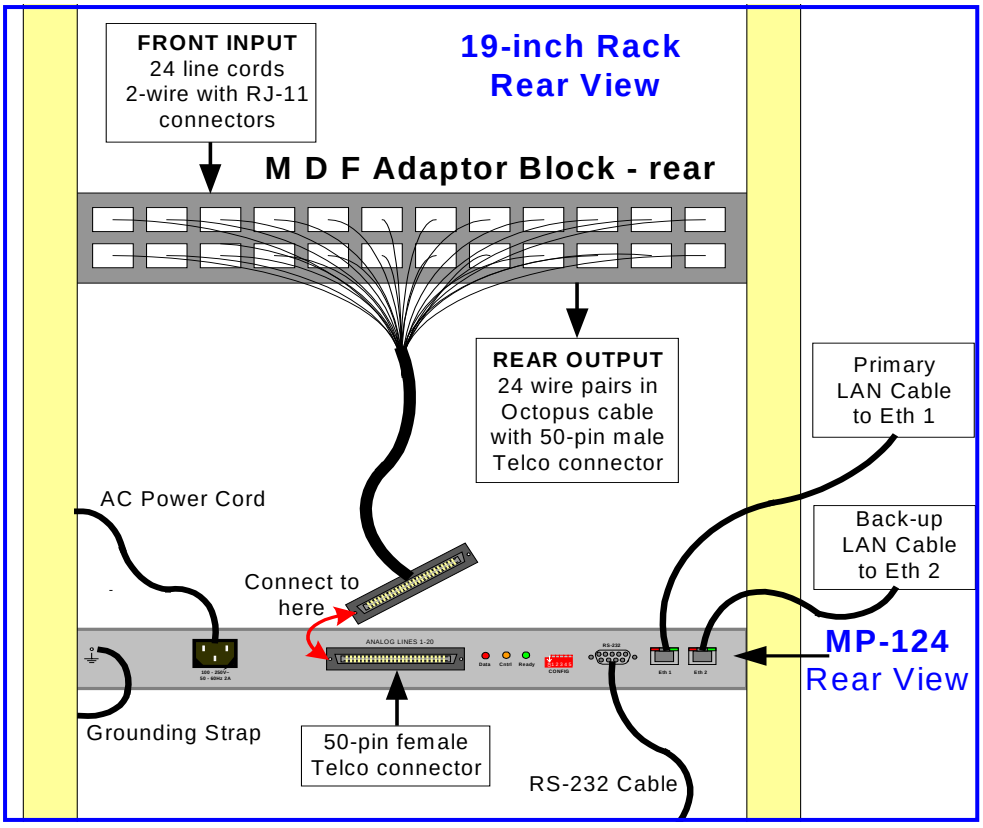
➢ **To unpack the MP-124:**

- Open the carton and remove packing materials
- Remove the **MP-124** from the carton
- Check that there is no equipment damage
- Check, retain and process any documents
- Notify AudioCodes of any damage or discrepancies
- Retain any diskettes or CDs

3.1.2 MDF Adaptor

To connect 24 2-wire lines into the **MP-124**, a Main Distribution Frame (MDF) Adaptor Block can be used as shown in Figure 3-1. This converts the standard RJ-11 connectorized phone line into a plain pair of wires that are terminated within a 50-pin Telco connector.

Figure 3-1: MP-124 in a 19-inch Rack with MDF Adaptor



Note: The only equipment shown in Figure 3-1 and supplied by AudioCodes is the **MP-124** H.323 Gateway and as an option, the MDF Adaptor.

As **input** (on the front of the 19-inch rack), the Adaptor Block takes in 24 2-wire lines with standard RJ-11 connectors.

As **output** (on the rear of the 19-inch rack), the Adaptor Block provides 24 wire pairs, which need to be terminated into a single 50-pin male Telco connector.

The 50-pin connector **must** be wired according to the pinout in Table 3-1 and Figure 3-2, shown below.

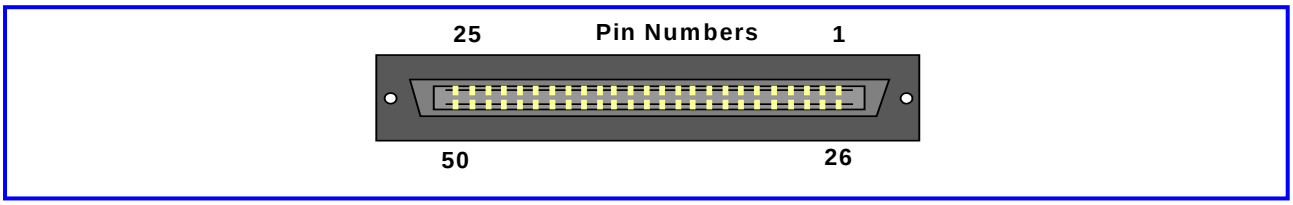
3.1.3 Cable Connections

The 50-pin Telco connector mounted on the rear of the **MP-124** is wired according to the pinout in Table 3-1 and Figure 3-2, shown below. The User's cable-mounted 50-pin Telco connector, supporting the 24 2-wire phone lines, **must** be wired identically.

Table 3-1: Pin Allocation in 50-pin Telco Connector

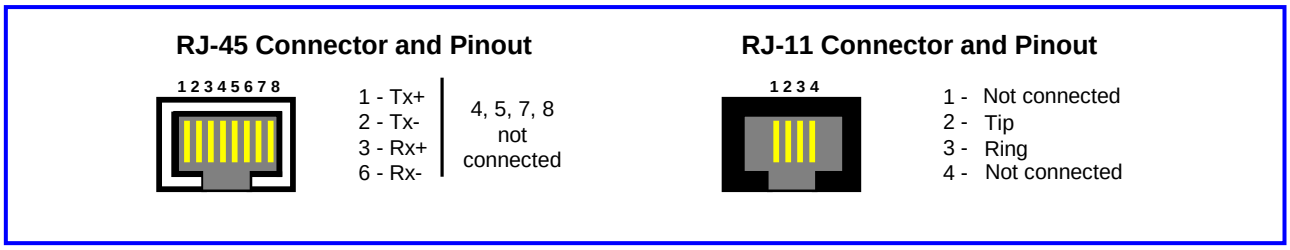
Phone Channel	Connector Pins	Phone Channel	Connector Pins
1	1/26	13	13/38
2	2/27	14	14/39
3	3/28	15	15/40
4	4/29	16	16/41
5	5/30	17	17/42
6	6/31	18	18/43
7	7/32	19	19/44
8	8/33	20	20/45
9	9/34	21	21/46
10	10/35	22	22/47
11	11/36	23	23/48
12	12/37	24	24/49

Figure 3-2: 50-pin Telco Connector

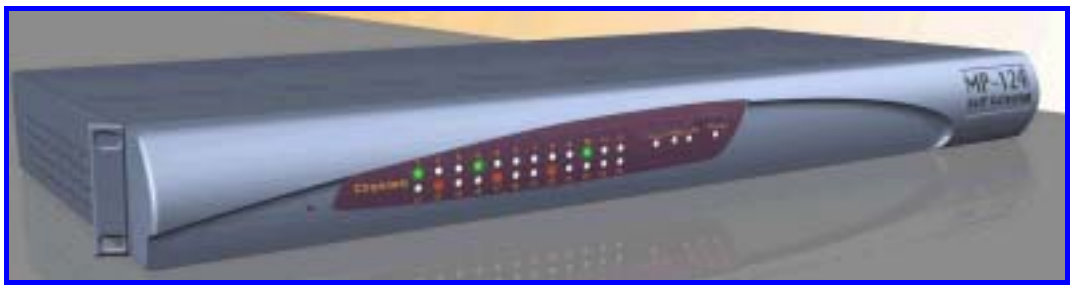


The RJ-45 (Ethernet) and RJ-11 (POTS) pinouts and connectors are shown in Figure 3-3. Pins are numbered from the left with the latching finger position at the bottom.

Figure 3-3: RJ-45 and RJ-11 Connectors and Pinouts


RJ-45 Connector and Pinout
1 2 3 4 5 6 7 8
1 - Tx+
2 - Tx-
3 - Rx+
6 - Rx-
4, 5, 7, 8 not connected
RJ-11 Connector and Pinout
1 2 3 4
1 - Not connected
2 - Tip
3 - Ring
4 - Not connected

3.1.4 19-inch Rack Mounting for MP-124



MP-124 Gateway is supplied with brackets ('ears') fitted to each side of the enclosure so that the **MP-124** can be immediately installed in the 19-inch rack.

➢ To install the rack mount MP-124, take the next 9 steps:

Step 1 Insert the **MP-124** into the 19-inch rack, adjust it to the correct position and use two standard rack-screws (not supplied) to secure each of the two brackets to the rack frame.

Step 2 Insert each of the RJ-11 connectors on the 2-wire line cords of the POTS phones into the RJ-11 sockets on the front of the MDF Adaptor Block

Up to 3,000 m (10,000 feet) of 24 AWG line cord can be used to connect telephones.

Step 3 Attach to the each of the sockets on the rear of the MDF Adaptor Block one pair of wires from a 25-pair Octopus cable.

Step 4 Connect the wire-pairs at the other end of the Octopus cable to a male 50-pin Telco connector. The pinout must be that shown in Table 3-1 and Figure 3-2 on page 24.

Step 5 Insert and fasten this 50-pin connector into the female 50-pin Telco connector mounted at the rear of the **MP-124** and labeled Analog Lines 1-24.

Step 6 Insert the RJ-45 connector of the 10/100 Base-T cable into the RJ-45 connector mounted at the rear of the **MP-124** and labeled Eth 1 for connection to your LAN.

Step 7 Connect an electrically grounded strap to the chassis ground screw on the rear of the **MP-124** and fasten it securely according to the current standards.

Step 8 Connect an electric power cord of the correct rating, from a grounded supply of the correct voltage, into the power socket mounted at the rear of the **MP-124** and labeled 100 - 250 V ~ 50 – 60 Hz 2A.

Step 9 Observe the front panel LEDs to determine the functioning of the **MP-124**.

The Channel LEDs indicate that the telephones connected to the rear 50-pin connector are each in one of the following states:

a. ringing or in the Off Hook position Green
b. normal operation Blank
c. not functioning Red

The functions of all the LEDs of the **MP-124** are shown in Table 3-2 on page 26.

3.2 Front Panel LED Indicators

Figure 3-4: Front Panel LED Indicators

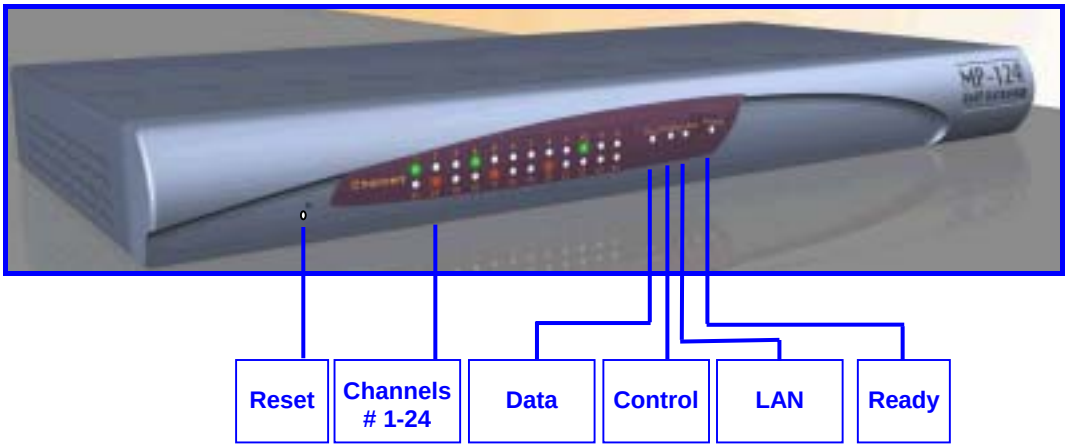


Table 3-2: Function of Front Panel LED Indicators

Label	Type	Color	State	Function
Data	Packet Status	Green	Blinking	Transmitting RTP Packets
		Red	Blinking	Receiving RTP Packets
		Blank	-	No traffic
Control	Control Link	Green	Blinking	Currently not implemented
		Red	ON	Currently not implemented
		Orange		Currently not implemented
LAN	Ethernet Status	Green	ON	Valid link to 10/100 Base-T hub/switch
		Red	ON	Malfunction
Ready	Device Status	Green	ON	Device Powered and Self-test OK
		Orange	Blinking	Software Loading/Initialization
		Red	ON	Malfunction
Channels # 1- 24	Tel Port	Green	ON	Off-Hook/Ringing for FXS Phone Port
	Tel Status	Red	ON	Malfunction
		Blank	-	Normal

3.3 Rear Panel LED Indicators/Connectors

Figure 3-5: Rear Panel LED Indicators and Connectors

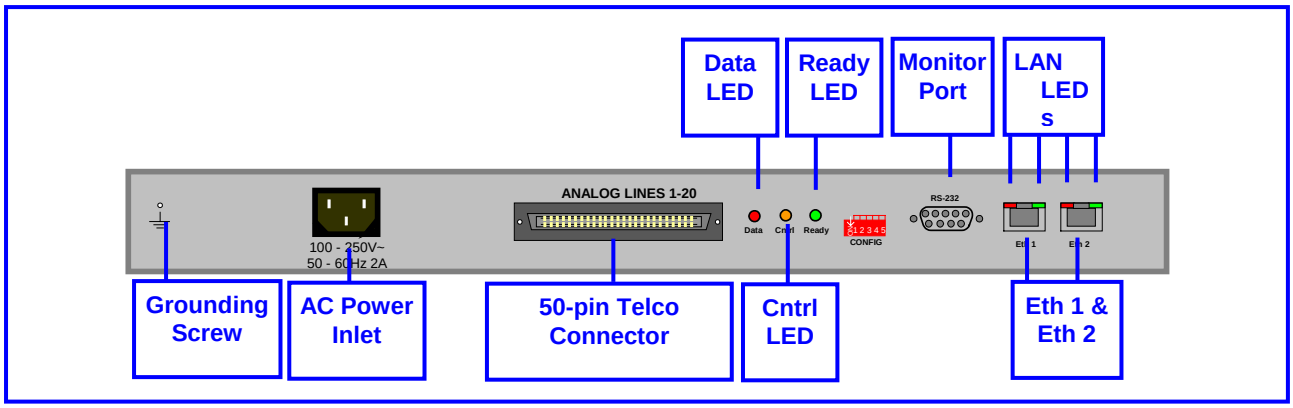


Table 3-3: Function of Rear Panel LED Indicators

Label	Type	Color	State	Function
Data	Packet Status	Green	ON	Transmitting RTP Packets
		Red	ON	Receiving RTP Packets
		Blank	-	No traffic
Cntrl	Control Link	Green	ON	Currently not implemented
		Red	ON	Currently not implemented
		Orange	ON	Currently not implemented
Ready	Device Status	Green	ON	Device Powered and Self-test OK
		Orange	ON	Software Loading/Initialization
		Red	ON	Malfunction
Eth 1	Ethernet Status	Green	ON	Valid link to 10/100 Base-T hub/switch
		Red	ON	Malfunction
Eth 2	Ethernet Status	Green	ON	Valid link to 10/100 Base-T hub/switch
		Red	ON	Malfunction

Table 3-4: Function of Rear Panel Connectors/Switches

Label	Type	Function	Comment
100-250 V~ 50 - 60 Hz 2A	3-pin AC	AC input	Connection to AC power cord
⏚	Grounding Screw	Chassis ground	
Analog Lines 1 to 24	50-pin Telco connector	FXS Ports	2-wire Loop Start interface
Eth 1	RJ-45	10/100 Base-T	Shielded port to Ethernet LAN. This is the default port.
Eth 2	RJ-45	10/100 Base-T	Shielded port to Ethernet LAN. The port is not in use for current SW release.
RS-232	DB-9, DCE	Status Messages	Gateway connects to PC's RS-232 port with a straight cable (refer to Figure 8-1 on page 100)..)

Note: The DIP switch located on the **MP-124** rear panel is not functional and should **not** be used.

4 Software Installation

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS Gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO Gateways.

4.1 Installation Package

The Installation Package includes the following Software and Utilities:

Software:

MP108_ram.cmp	- Image Software for download to **MP-10x/FXS** and **MP-10x/FXO** Gateways
MP124_ram.cmp	- Image Software for download to **MP-124** Gateway
usa_tones.dat	- Call progress tones *dat* file for download
usa_tones.ini	- Call progress tones *ini* file (used to create *dat* file)
H323gw_FXS.ini	- *ini* example file for FXS **MP-1xx** Gateways
H323gw_FXO.ini	- *ini* example file for FXO **MP-10x** Gateways
MP1xx_Coeff_FXS.dat	- Telephony interface configuration file for **MP-1xx/FXS** Gateway
MP10x_Coeff_FXO.dat	- Telephony interface configuration file for **MP-10x/FXO** Gateway
MIB library	- Library of SNMP MIBs

Utilities:

ini file utility.xls	- Excel™ utility for creation of ini file
TPDMUtil.exe	- Call progress tones file generator utility
Bootp_install.exe	- AudioCodes BootP & TFTP configuration utility

4.2 MP-1xx Initialization

MP-1xx Gateway comes with pre-installed software. The basic installation can be done using AudioCodes configuration utility, or from Web browser, such as from Microsoft Internet Explorer.

To change network parameters, use Web browser, AudioCodes BootP & TFTP configuration utility or use third party BootP server.

For setting the H.323 parameters in the *ini* file, either edit the *ini* example file, or generate such a file using the Excel utility provided.

The *ini* file and other configuration files can be downloaded directly from the Web browser using either HTTP protocol, or AudioCodes-provided configuration utility or any standard TFTP server. The Image software file: *ram.cmp* is only used for software upgrade.

The Call Progress tone file *usa_tones.ini* is used to define call progress tone levels and their frequency. To change the tone's parameters, first modify the file, and then using "TPDMUtil.exe"

utility, convert the text *ini* file to binary *dat* file. This procedure is described in Section 6.4.

The Coeff_FXS.dat and Coeff_FXO.dat files can be used respectively to modify the **MP-1xx/FXS** and **MP-10x/FXO** telephony interface characteristics, such as DC and AC impedance, feeding current and ringing voltage. For more information, refer to Section 6.5

4.3 Quick Setup Procedure

The following procedure describes how to setup the **MP-1xx** Gateway with basic parameters using standard Web browser (such as Microsoft Explorer). It is assumed that the IP address of the Gateway is known. If the IP address is unknown, use AudioCodes Configuration utility (or any standard BootP server) to set the Gateway IP address and subnet mask.

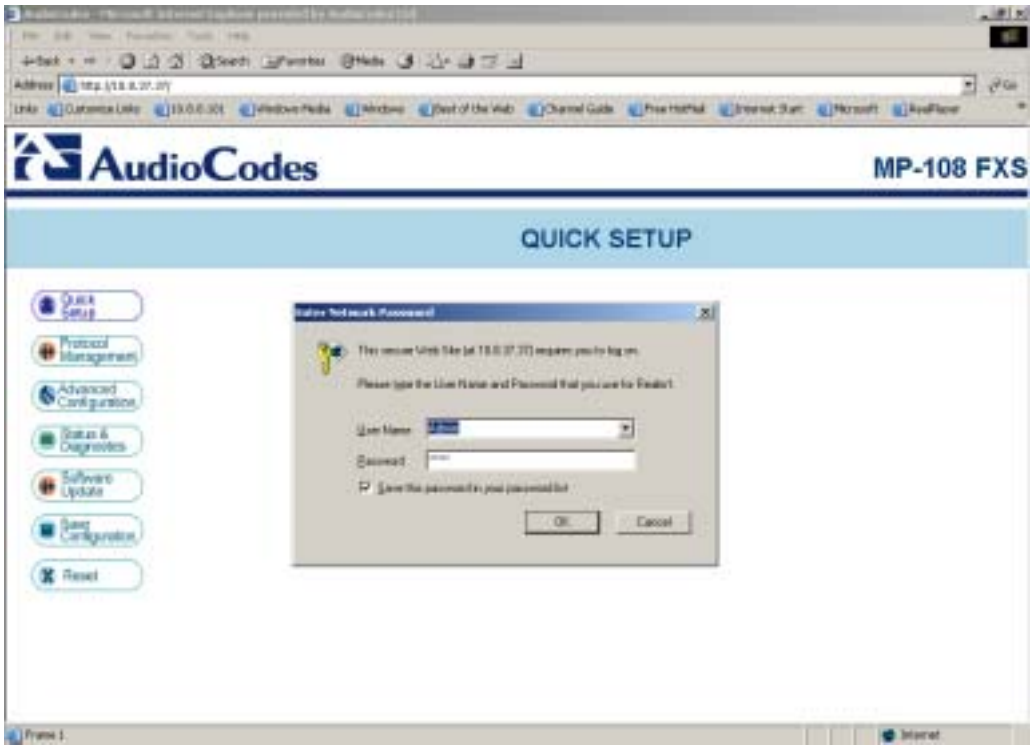
Usually the **MP-1xx** Gateways are shipped with the following network parameters:

- **MP-1xx** FXS Gateway IP address: 10.1.10.10
- **MP-10x** FXO Gateway IP address: 10.1.10.11
- Subnet: 255.255.0.0
- Default Gateway: 0.0.0.0

➢ For quick MP-1xx setup, take the next 12 steps:

1. Power-up the Gateway; after self-testing, in about 20 seconds the Ready LED on the front panel turns to green. Any malfunction changes the Ready LED to red.
2. Before using your Web browser to access the **MP-1xx**'s Embedded Web Server, change your PC's IP address and Subnet Mask to correspond with the **MP-1xx**'s factory default IP address and Subnet Mask.
3. Open any standard Web-browsing application such as Microsoft™ Internet Explorer™ (Ver. 5.0 and higher) or Netscape™ Navigator™ (Ver. 7.0 and higher) and specify the IP address of the Media Gateway in the 'Address' field (e.g., http://10.1.10.10 for FXS); the Embedded Web Server 'Enter Network Password' screen appears, Figure 4-1.

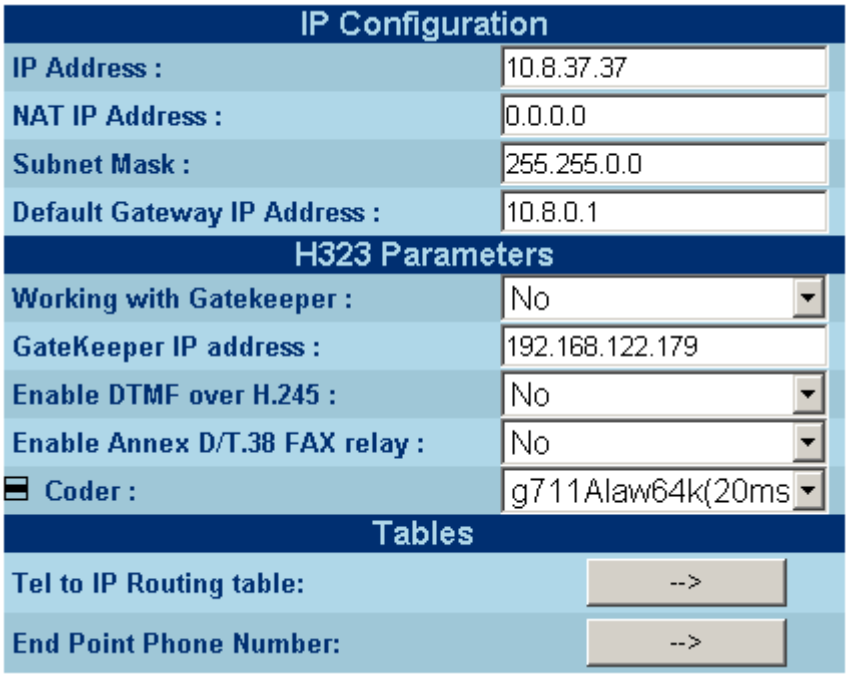
Figure 4-1: Web Browser Screen



4. Enter the User Name and Password (default: Admin, Admin). Note that the User Name and Password fields are case-sensitive. Click OK; the 'Quick Setup' screen is accessed.

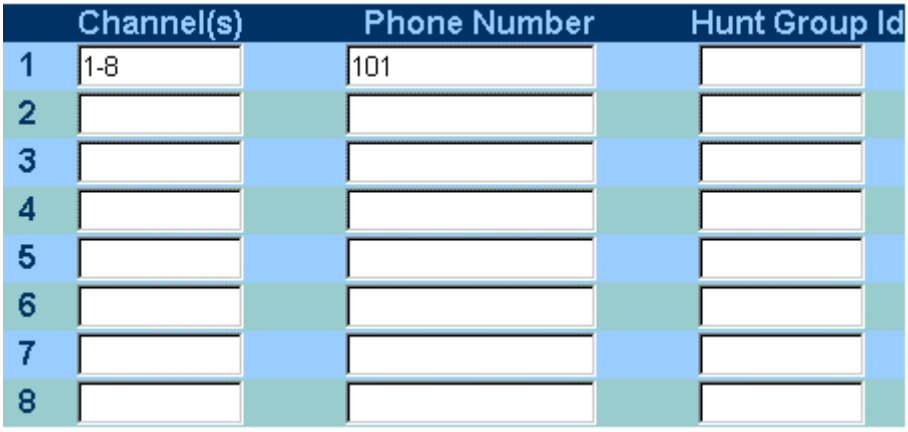
5. In the 'Quick Setup' screen set the Gateway's new IP Address, NAT IP Address, Subnet Mask and Default Gateway IP Address.
6. Set the basic H.323 Gateway Parameters, as shown in Figure 4-2 below.

Figure 4-2: H.323 Quick Setup



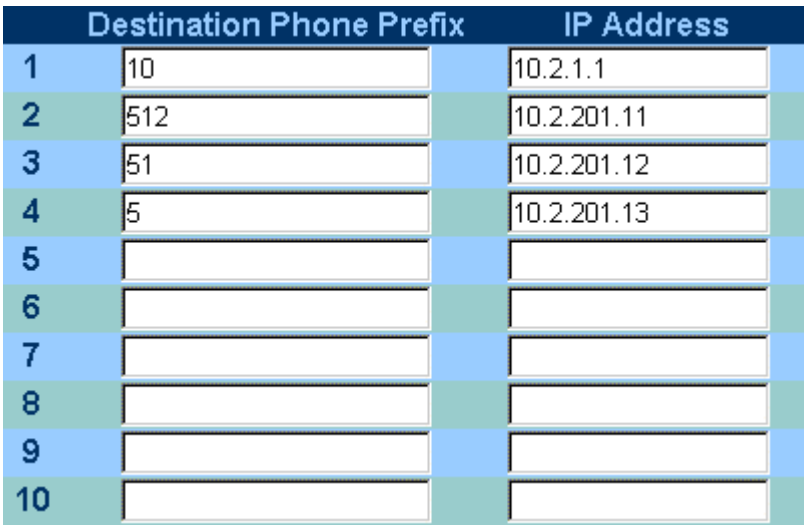
7. Open the "End Point Phone Number" table and fill in the Gateway phone numbers. For example, for **MP-108**, enter "1-8" in 'Channel(s)' field and starting Phone number, such as 101 in adjacent 'Phone number' field. The **MP-108** physical ports are associated with phone numbers 101 to 108. **You don't need to fill the 'Hunt Group ID' column.** Click the Submit button and close the window.

Figure 4-3: Endpoints' Phone Numbers



8. When working without a Gatekeeper the "Tel to IP Routing" table needs to be defined: Fill in the "Destination phone prefix" (prefix of called number) and the associated IP address. In the following example, an outgoing (Tel->IP) call with called number starting with 512 is sent to IP address 10.2.201.11; while all other numbers starting with 51 are sent to IP address 10.2.201.12. Assuming that the Gateway IP address is 10.2.1.1, all local calls starting with "10" (such as 101-108) are routed back to the Gateway. When Gatekeeper is used, the routing table can be used optionally as fallback in case of failure. Click the submit button and close the window.

Figure 4-4: Tel to IP routing table



	Destination Phone Prefix	IP Address
1	10	10.2.1.1
2	512	10.2.201.11
3	51	10.2.201.12
4	5	10.2.201.13
5		
6		
7		
8		
9		
10		

9. Click the 'RESET' button and click 'OK' in the prompt; wait approx. 60 seconds and refresh the Web page
10. Restore your PC's IP address and Subnet Mask to their original values.
11. Using "Advanced Configuration" and "Protocol Management" screens you can modify all H.323 and other Gateway parameters. Refer to Section 7.2 for detailed directions of operating under Web management control.
12. Finally you can establish phone calls between phones connected to the Gateway ports (for FXS Gateway) by dialing 101-108 numbers.

4.4 BootP and TFTP Procedures

If the Gateway IP address is known, you can use Web browser control for Gateway configuration and provisioning. Otherwise, you can use BootP (or DHCP) and TFTP protocols for initialization and software download.

Each time the **MP-1xx** Gateway is powered-on, it performs the standard BootP procedure.

If "DHCPEnable = 1" line is included in Gateway's *ini* file and if the BootP server was not found, the Gateway initiates standard DHCP procedure to configure the Gateway network parameters (IP address, Subnet mask and Default router address). If DHCP procedure is used, you also need to find the new Gateway IP address allocated by DHCP server. Usually this information can be provided by System Administrator or you can connect the MP-1xx to PC hyper terminal using RS-232 cable and view the Gateway network settings after the Gateway is reset (see chapter 8.4).

If the BootP/DHCP server has not been found, the **MP-1xx** Gateway starts working from its internal flash memory.

Usually the application software already resides in the **MP-1xx** flash memory; therefore there is no need to use the BootP or TFTP procedure. Their download need only be used for changing the **MP-1xx** configuration or for a new software upgrade.

The BootP Protocol enables the network administrators to manage the configuration of the **MP-1xx** Media Gateway from a central configuration server - BootP/DHCP server.

The following RFCs (IETF Requests for Comment) describe BootP in detail: RFC 951, RFC 1542 and RFC 2132.

Downloading of the image file by the **MP-1xx** is performed using Trivial File Transfer Protocol (TFTP). TFTP protocol is described in RFC 906 and RFC 1350.

Although DHCP and BootP servers are very similar in operation, the DHCP server includes some differences that might prevent its operation with BootP clients. However, many DHCP servers, such as Windows™ NT DHCP server, are backward-compatible with BootP protocol and can be used for **MP-1xx** configuration.

Note: The BootP server is normally used to configure the **MP-1xx** initial parameters. Once

this information has been provided, the BootP server is no longer needed. All parameters are stored in non-volatile memory and used when the BootP server is not accessible. The BootP server is required again if, for example, the **MP-1xx** IP address is to be changed.

Using BootP procedure, the following parameters are configured:

- Boot & *ini* File Names — Optional, refer to Note 1 below.
- Local IP Address — IP address of your **MP-1xx** Gateway.
- Gateway IP Address — If Default Gateway/Router is required, otherwise enter 0.0.0.0 address.
- Subnet Mask — Subnet Mask of your **MP-1xx** Gateway.
- TFTP Server IP Address — Refer to Note 2 below.

Note 1: Boot file name can contain one or two file names. ram.cmp file name to be used for download of application image and mp1xx.*ini* file name to be used for **MP-1xx** provisioning. Either one, two or no file names can appear in the Boot file name field. To use both file names use ";" separator (without blank spaces) between the xxx.cmp and the *yyy.ini* files (e.g., *ram.cmp;H323gw.ini*).

Note 2: Usually TFTP and BootP servers are installed on the same Host. However, when using AudioCodes Configuration utility or Microsoft™ DHCP server, it is possible to set the IP address of TFTP server (Boot Server Host Name field), and in this case BootP and TFTP servers can run from different Hosts.

4.4.1 Configuring the TFTP Server

➢ To configure the TFTP Server, take the next 4 steps:

1. Set the default directory where the image file resides (C:\AudioCodes\…).
2. Copy the Image file (such as ram.cmp to the TFTP default directory on your Host PC.
3. Copy the *ini* file and other optional configuration files (Call Progress tones and Coeff.dat files) to the TFTP default directory on your Host PC. Ensure correct coeff.dat file is used. Two different coefficient files are provided, for **MP-1xx/FXS** and **MP-10x/FXO**.
4. Set the TFTP timeout to 3 seconds and number of retransmissions to 20.

4.4.2 Using AudioCodes BootP/TFTP Configuration Utility

The AudioCodes Configuration utility provides an easy way to configure the **MP-1xx** Gateway. Similar to other BootP and TFTP servers, the application can be installed on Windows™ 98 or Windows™ NT/2000. With AudioCodes' BootP/TFTP Server Configuration utility, it is possible to use the integrated TFTP server (part of the Utility) or to install TFTP server on a different host. The utility enables remote reset of the **MP-1xx** unit for triggering the initialization procedure (BootP & TFTP). For details of the procedure, refer to Appendix A - BootP/TFTP Configuration Utility.

4.4.2.1 Configuration Utility Main Features

- BootP server supporting hundreds of entries.
- Integrated TFTP server.
- Common Log window displaying BootP and TFTP status.
- Contains all data required for provisioning of AudioCodes products.
- Provides the TFTP server address, enabling network separation of TFTP and BootP servers.
- Tools for backup and for restoring the local database.

- Templates.
- User-defined names for each entity.
- Protection against entering fault information.
- Unicast or Broadcast BootP response.

4.4.3 Configuring the Windows™ NT DHCP Server

If Microsoft™ Windows NT DHCP server is used in your organization, the server can be used in reservation mode to provide an IP address and other necessary information to the **MP-1xx** Media Gateway.

To configure the Microsoft™ Windows™ NT DHCP Server to assign IP address information to BootP clients, add a reservation for each BootP client.

For information about how to add a reservation, view the "Managing Client Reservations Help" topic in DHCP Manager.

The reservation builds an association between the media access control address (12 digits, provided in **MP-1xx** documentation) and the IP address. Windows™ NT Server provides the IP address based on the **MP-1xx** media access control (MAC) address in the BootP request frame.

To configure the Microsoft™ Windows™ NT DHCP server to provide boot file information to BootP clients, edit the BootP Table in DHCP Manager. The BootP Table is located in the Server Properties dialog box that can be accessed from the Server menu. For information about how to edit the BootP Table, view the "BootP Table" Help topic in DHCP Manager.

The following parameters must be entered:

- Local IP address — IP address of your **MP-1xx** Gateway.
- Subnet mask — Refer to Section 4.4 Note 2, for the mask limits.
- Gateway IP address — Default Gateway IP address
- Boot File name — Optional, refer to following Note.

Note: Boot file name normally should not be used. This field is only used for software upgrade (refer to Section 4.5.

Refer to Utilities User Manual for detailed description of configuration of Windows NT DHCP server.

4.4.4 Other TFTP & BootP Servers

Third party TFTP and BootP servers can be used; for example Weird Solutions™ (www.weird-solutions.com).

4.5 MP-1xx Software Upgrade

4.5.1 General Upgrade Procedure

MP-1xx includes on-board flash memory already programmed with application software. The following procedure replaces the old stored software with the new version. To run this procedure BootP and TFTP servers are required. Web browser can be used instead of BootP server (refer to Section 7.2).

Note1: The file extensions *cmp* and *ini* should be written in lower case letters.

Note2: When upgrading the internal software (either from a TFTP Server or from the Embedded Web Server) it is mandatory to download the ram.cmp file and all other related files: ini, coefficient and call progress tone.

➢ To upgrade the integral Software take the next 5 steps:

1. Start the TFTP and BootP servers, refer to Appendix A.
2. Copy the new *ramxxx.cmp* file, *mp108.ini* file (e.g., *H323gw.ini*) and optional configuration files to the default TFTP server directory.
3. Set the Boot file and *ini* file names in the Web browser Network settings page or in the BootP server: *ramxxx.cmp -fb;mp108.ini*. Other network parameters stay unchanged (IP address, subnet mask,…). If so required, it is possible to update only the *mp108.ini* parameters. For this option, set the boot file name to: *mp108.ini* (without preceding *ramxxx.cmp*). After **MP-1xx** power reset, the *ini* parameters are downloaded using the TFTP procedure and stored in the non-volatile memory.
4. Reset the **MP-1xx**. Wait about 20 seconds until the Ready LED changes to Green.
5. After accomplishing BootP and TFTP procedures, the new software is downloaded and stored in the **MP-1xx** unit's flash memory.

Note: The parameter "-fb" added to Boot file name is used to specify the burning of flash memory with new software image. To test new software version without replacing the old version, skip the "-fb" parameter. In this case, the new software is downloaded directly to RAM, and not permanently stored into flash.

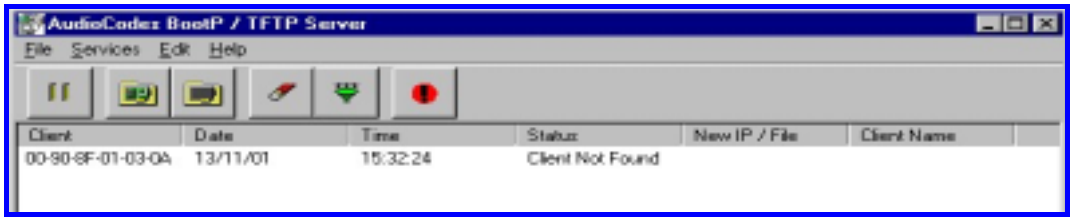
4.5.2 Upgrade Procedure Using AudioCodes Configuration Utility

The following procedure describes how to upgrade the **MP-1xx** software using the AudioCodes configuration utility.

➢ To upgrade the Software using the AudioCodes Configuration Utility take the next 13 steps:

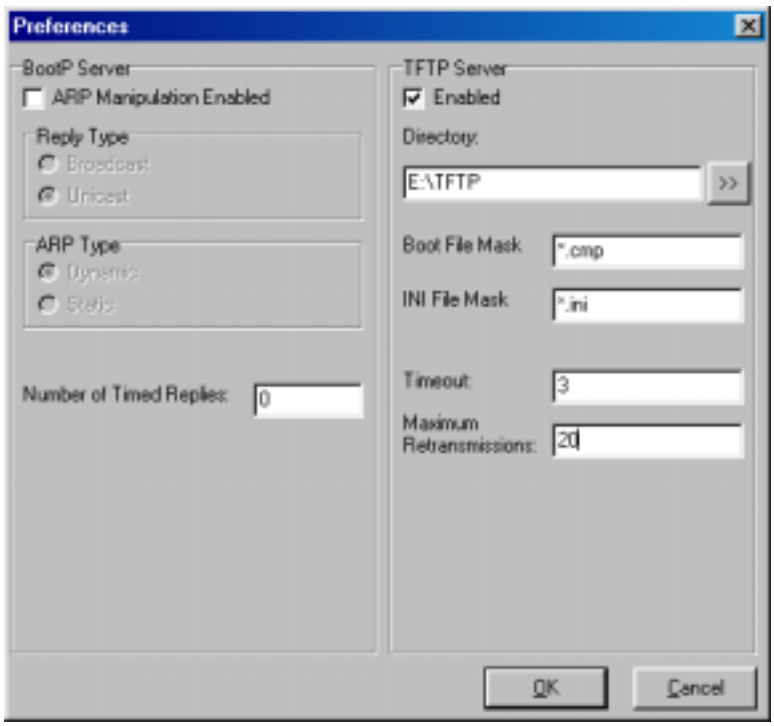
1. Install AudioCodes' Configuration utility from the AudioCodes Software CD, Document # LSTC00005 (MediaPack Series).
2. Open the AudioCodes Configuration utility from Start>Programs>BootP; the AudioCodes Configuration utility main screen opens:

Figure 4-5: AudioCodes Configuration Utility Main Screen



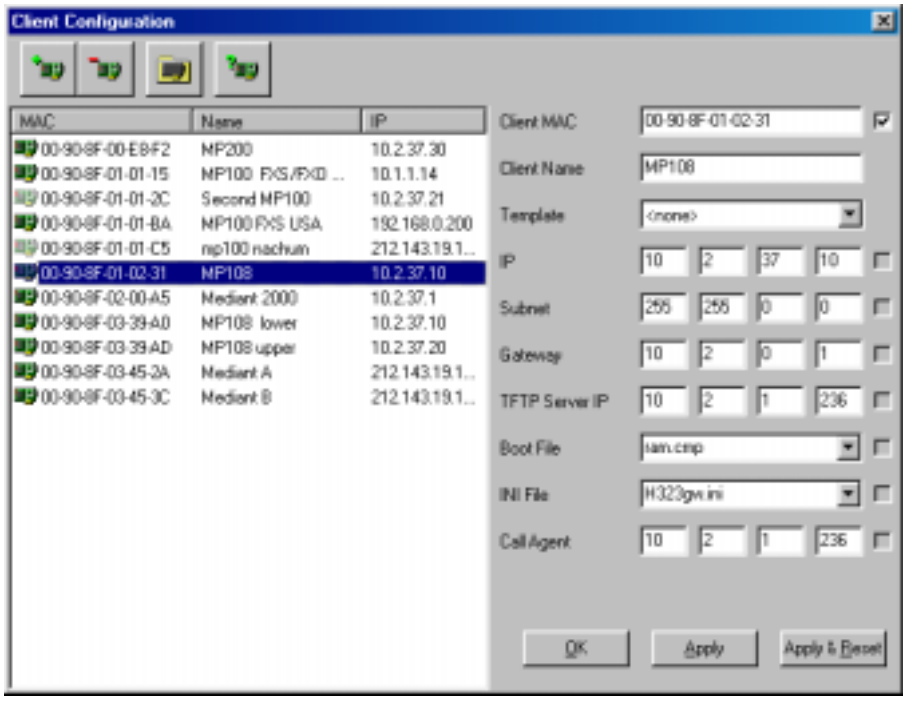
3. Click on the Edit tab to open the Edit menu.
4. Select Preferences to open Preferences window shown below 36.

Figure 4-6: Preferences Screen



5. In the Directory field, click on the >> button and navigate to the directory of the source **.cmp* and **.ini* files. All downloaded files should reside in this folder, including ram.*cmp*, mp108.*ini*, Coeff.dat and Call Progress tone.*dat* files.
6. Click OK to return to the main screen.
7. In the Services menu, choose Clients. This opens the Client Configuration screen shown in Figure 4- on page 36. The parameter fields displayed on the right side of the screen constitute the **MP-1xx** software profile configuration. The parameter fields are all blank in the case of a Client Not Found.

Figure 4-7: Client Configuration

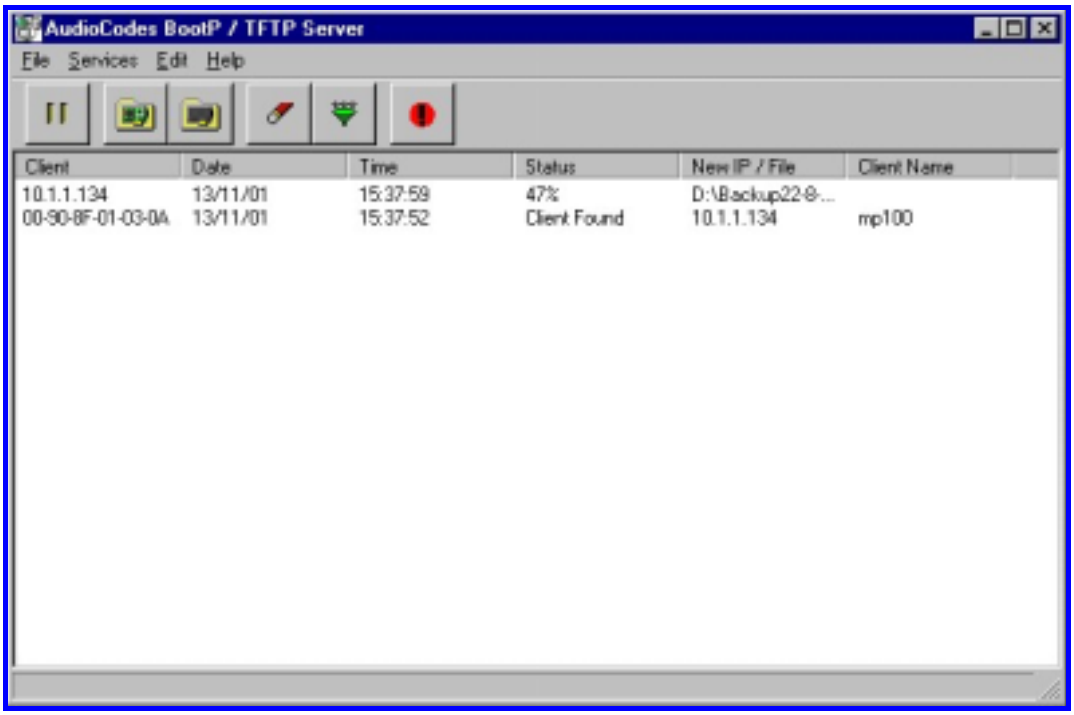


8. Enter the Client MAC address and Client Name.

9. Enter the IP address (such as 10.2.37.1).
10. Enter the Subnet (such as 255.255.0.0); set the Subnet to a valid value in accordance with the IP address.
11. Enter the IP address of the default Gateway; it can be any address within the subnet.
12. Select the required Boot and *ini* Files
13. To permanently store the new image file in the **MP-1xx** flash memory, add –fb suffix to Boot file name, such as "ram.cmp –fb". After entering the file names, click "Apply & Reset" button.

The following status messages is displayed in the AudioCodes BootP/TFTP Server main screen:

Figure 4-8: AudioCodes Configuration Utility – TFTP download


AudioCodes BootP / TFTP Server
File Services Edit Help
Client Date Time Status New IP / File Client Name
10.1.1.134 13/11/01 15:37:59 47% D:\Backup22-9...
00-90-8F-01-03-0A 13/11/01 15:37:52 Client Found 10.1.1.134 mp100

Reader's Notes

5 Profiling & Operation

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS Gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO Gateways.

5.1 H.323 Profile

5.1.1 Supported H.323 Features

The **MP-1xx** Gateway implements RadVision™ H.323 version 4.0 protocol stack. In this version, the Gateway features the following:

Gatekeeper:

- Can work without Gatekeeper (use internal phone table).
- Can register to known Gatekeeper.
- MP-1xx can function as an H.323 Gateway or it can imitate an H.323 terminal with multiple aliases (e.g., up to eight for **MP-108**).
- Supports GK registration with prefixes
- Can use routed-mode calls.
- Can use direct-mode calls.
- Redundant Gatekeepers can be defined in *ini* file
- Fallback to internal routing table if no communication with GK
- "TimeToLive" parameter is supported. The **MP-1xx** Gateway sends Registration requests up to "TimeToLive" expiration.
- IRR messages for KeepAlive are supported.
- Supporting mapping of destination (Alias) number in ACF message by Gatekeeper.
- Supporting RAI (Resource Available Indication) messages, informing gatekeeper that the Gateway resources are below a threshold.

Call setup:

- Can use Normal Connect procedure.
- Can use Fast Connect procedure with or without opening immediately H.245 channel.
- Can use Tunneling.
- Can negotiate codec from a list of given codecs for Normal or Fast Connect procedures.
- Optional opening of H.245 channel when using Fast Connect.

Other:

- Country Code (0xB5) and Manufacturers Code (0x28) are used in H.323 messages
- Support H.323 Annex D, T.38 real time FAX.
- Supports H.450 Call Hold, Call Transfer and Call Forwarding supplementary services (H.450.1, H.450.2, H.450.3 and H.450.4).
- Supported codecs: G.711 A-law (5, 10 and 20 msec), G.711 μ-law (5, 10 and 20 msec), G.723 (5.3 and 6.3 kbps), G.729 (8 kbps, 10 and 20 msec), G.726 (16, 32 kbps, 20 msec), NetCoder (6.4, 7.2, 8.0 and 8.8 kbps, 20 msec).
- Supports DTMF and HookFlash signal out of band using H.245 channel (using "Alphanumeric" field).
- Supports DTMF and HookFlash signal out of band using H.225/Q.931 keypad facility messages
- Support of one or two stage dialing for network to PBX/PSTN calls, using **MP-10x/FXO** Gateway.
- Support of collecting Digits from PBX by **MP-10x/FXO** Gateway.
- Gateway registration with prefixes (applicable to FXO Gateway)
- Reopening of logical channel and implementation of empty terminal capability set.
- Configurable H.323 Port Range.
- H.225/Q.931 Progress Indicator parameter is supported for Fast Connect, enabling playing of local ringback tone or to cut through the voice channel to listen to remote call progress tones/messages.
- Supports detection of Caller ID signal (Bellcore and ETSI standards) from PBX line by **MP-10x/FXO** Gateway, and interworking it to H.323 network.

For detailed information, refer to the latest **MP-1xx**/H.323 Release Notes, Document #: LTRT-00652.

5.2 Using H.323 Gateway Features

Details of how to use and configure the H.323 Gateway Parameters (from *ini* file) are shown below in Table 5-1.

Table 5-1: Using H.323 Gateway Features (continues on pages 41 to 44)

Feature	Parameter	Sheet of Excel Utility	Value
Use given Gatekeeper	IsGatekeeperUsed	H323gw	1
	GatekeeperIp Single IP address or several IP address if redundant gatekeeper option is enabled.	H323gw	IP
Redundant Gatekeepers	IsRedundantGKUsed Specify list of gatekeeper IP addresses using GatekeeperIp parameter. Gateway starts working with first GK from the list. In case of a failure with first GK, the Gateway tries to find other GKs from the list. Once GK is found, the Gateway continues working with it, until the next failure. If none of the GKs respond, Gateway goes over the list again.	H323gw	1
No Gatekeeper,use routing table	IsGatekeeperUsed	H323gw	0
	Define routing table using: a. PREFIX = prefix, <IP address> list b. Prefix = *, <IP address>	Routing or/& Prefix Tables	Phone numbers, Prefixes and IPs
	When Gatekeeper isn't used, it is necessary to define IP routing table, to enable **MP-1xx** Gateway to find destination IP address, according to received dial number. The routing table is defined in i*ni* file, using PREFIX or per phone number definitions. The Gateway first searches for Phone table to find a destination IP address, than it looks for Prefix parameter, and later for "Prefix = *,<IP address>" definition. "Prefix = *, <IP address>" defines destination IP address for any other phone number.		
Fall Back to internal routing table	IsGKFallbackUsed Gateway starts using internal routing table (phone to IP, or prefix to IP) if communication with Gatekeeper is lost. Gateway returns to Gatekeeper when communication is restored.	H323gw	1
Set numbers to end points	Channel2Phone= <channel>,<phone> or ChannelList = port, number, phone	EndPoints	Phone numbers
	The easiest way to define endpoint phone number is to use ChannelList parameter. For example, to define 101 to 107 numbers for an **MP-108**, use a single line: ChannelList = 0,8,101. The first parameter (0) indicates the first endpoint number. The second parameter (8) indicates the number of endpoints. The third parameter (101) indicates the first endpoint phone number. Up to ten such ChannelList definitions can appear in the same *ini* file. One or more phone numbers in the ChannelList can be modified by using Channel2Phone definition, following the ChannelList parameter in the *ini* file.		
Dial plan	MaxDigits Maximal number of digits in dialed number.	General	Max digits
	TimeBetweenDigits Timeout between dialed digits, used to terminate dialing. Usually it is set to 4 seconds.	General	Time in seconds
Use "Fast Connect"	IsFastConnectUsed	H323gw	1

Table 5-1: Using H.323 Gateway Features (continues on pages 41 to 44)

Feature	Parameter	Sheet of Excel Utility	Value
procedure			
Fast start and Tunneling can be used together			
Use "Tunneling" procedure	IsTunnelingEnabled	H323gw	1
Default number	DefaultNumber	General	Default number
	This is used to get call setups from end-points that do not send called phone number.		
Automatic dialing	IsDialNeeded	General	0
	TargetOfChannel	Automatic Dialing	Phone numbers to dial
	This is used to perform automatic dialing once OFF HOOK is detected in FXS Gateway or ringing is detected in FXO port. There is no need to dial in this mode. For each channel, define destination phone number, using "TargetOfChannel<channel> = phone number" definition.		
Choose Coder	CoderName	General	Preferred coder name
Several Coders	CoderName	General	List of coders
	In this mode, several codecs are sent in capabilities message. After negotiation with the remote side a single coder will be selected for transmit and receive.		
Annex D Fax	IsFaxUsed	Channel	1
	To use Annex D Fax, configure: UseT38orFRF11 = 1 (T.38) FaxTransportMode =1 (relay)	Fax	ports
	To improve Fax transmission over congested IP network, use optional redundancy parameters: FaxRelayEnhancedRedundancyDepth = N FaxRelayRedundancyDepth =M N = 0 to 4 M = 0 to 2	Channel	
One Stage Dialing, **IP → FXO calls**	IsTwoStageDial	H323gw	0
	IsUseFreeChannel	General	1
	MP-10x/FXO seizes the next available FXO line, and dials the destination phone number received in Setup message. Use the 'IsWaitForDialTone' parameter to specify whether the dialing comes after detection of dial tone, or immediately after seizing the line. **MP-10x**/FXO supports registration of prefixes to Gatekeeper. If your gatekeeper doesn't support the Gateway's prefix registration, you can probably configure the Gatekeeper to enable routing of Call with destination phone number to correct **MP-10x**/FXO IP address. Usually this is done by allocating telephone numbers range or prefix to **MP-10x** FXO IP address. The FXO Gateway releases the call if busy or fast busy (reorder) tone is detected on the FXO port.	General	

Table 5-1: Using H.323 Gateway Features (continues on pages 41 to 44)

Feature	Parameter	Sheet of Excel Utility	Value
Two Stage Dialing, **IP → FXO calls**	IsTwoStageDial IsUseFreeChannel	H323gw General	1 1
	For 'Two Stage Dialing' the **MP-10x**/FXO seizes the next available PSTN/PBX line, without performing any dial, the remote user is connected over IP to PSTN/PBX, and all further signaling (dialing and call progress tones) is done directly with the PBX without Gateway intervention. Usually the phone number received in Setup message is not used, however if 'IsUseFreeChannel = 0', the phone number received in Setup, is used for seizing specific FXO line that has same number.		
	Refer to the remote extension application note. The FXO Gateway releases the call if busy or fast busy (reorder) tone is detected on the FXO port.		
Using out of band DTMF signaling over H.245 and HookFlash signaling	IsDTMFUsed DTMFTransportType DTMFDetectionPoint Is931MsgUsed IsHookFlashUsed OpenH245onFS	H323gw Channel Channel H323gw H323gw H323gw	1 0 0 0 1 1
	When using H.245 out of band signaling, disable inband signaling (mute and don't relay the digits). In addition, disable the Q.931 sending. Using "DTMFVolume = " set the level of played DTMF digits. The HookFlash signal is detected at FXS endpoints and generated at FXO endpoints, according to "FlashHookPeriod = " parameter. Note: When using Fast Start the H.245 channel remains close, and out of band signaling can't be used, unless "OpenH245onFS" is set to 1 or remote Gateway/terminal opens the H.245 channel.	Channel H323gw	
Open media only on connect	IsFSOpenMediaONConnect	H323gw	0
	This feature is useful when Tunneling or Fast Connect are used. Voice channel in egress Gateway is opened immediately after sending an ALERT message. If this is not the appropriate procedure, the User can set this parameter to "1". to open voice channel only after CONNECT is sent.		
Using Hunt Groups	TrunkGroup_x = a-b, <starting number> IsUseFreeChannel PSTNPrefix = <prefix>, <Hunt Group ID> AddTrunkGroupAsPrefix This feature can be used to define group of Gateway ports, called hunt groups. For example: TrunkGroup_1 = 1-4,100 TrunkGroup_1 = 8,200 TrunkGroup_2 = 5-7,300	TG General TGRoute	range 1 0

Table 5-1: Using H.323 Gateway Features (continues on pages 41 to 44)

Feature	Parameter	Sheet of Excel Utility	Value
	To use the hunt groups it is required to define also "IsUseFreeChannel=1" , and to define routing rules, using "PSTNPrefix" parameter. The routing IP→TEL rules define per each called number prefix, the Hunt Group ID where the call will be send to. For example: PSTNPrefix = 101,1 Incoming IP→TEL call, with called number starting with 101 will select free channel from 1st Hunt group. Usually this feature will be used for FXO Gateways, but there are some hunt groups applications also with FXS Gateways. "**AddTrunkGroupAsPrefix = 1**" can be used when incoming Tel→IP call should be treated differently according to received hunt group. If this feature is enabled the Hunt group ID is added to called number as a prefix. The number manipulation and routing (Prefix=) rules will apply to the new destination number. **Note: When using Trunk groups don't use "ChannelList" or "Channel2Phone" parameters .**		

5.3 Alternate (Redundant) GK Implementation

1. The **MP-1xx** Gateway doesn't support Automate Gatekeeper discovery. Gatekeeper address is provided through initialization file (or via Web management). In AudioCodes, this procedure is referred to as "Redundant Gatekeeper" procedure. In general, it is identical to Alternate procedure described in H.323 standard, with the exception that the list of primary and alternate Gatekeepers is provided via initialization file.
2. In addition to primary Gatekeeper address, it is possible to provide list of two additional, "Alternate" Gatekeeper addresses, using the same initialization file.
3. Working with primary and alternate gatekeepers is according to procedure described in the H.323 standard (para. 7.2.6 Alternate gatekeeper procedures), with the exception that all Alternate Gatekeeper fields in RAS messages are ignored.
4. According to H.323, para 7.2.6 procedure, the **MP-1xx** Gateway supports the following:

- Endpoint attempts to register with the Gatekeeper using the list of Alternate Gatekeepers.
- If the endpoint is registered with the Gatekeeper and the Gatekeeper becomes unresponsive, the endpoint shall attempt to communicate with an Alternate Gatekeeper.
- If the endpoint determines that a temporary Alternate Gatekeeper is unresponsive, it attempts to retransmit the RAS request to another Alternate Gatekeeper. If all Alternate Gatekeepers are unresponsive to a RAS request, the endpoint assumes that the RAS request is rejected.

5. If an **MP-1xx** Gateway finds an alternative Gatekeeper from the list of alternate Gatekeepers, it registers with this Gatekeeper and continues working with it until next failure (current GK is not responding).
6. If all Gatekeepers are unresponsive, the **MP-1xx** Gateway tries again to establish communication, and to register to the first gatekeeper according to the provisioned Gatekeeper list.
7. The **MP-1xx** Gateway has an option to use internal phone to IP routing table (Fallback mode), when all Alternative Gatekeepers are unresponsive. If this option is enabled, **MP-1xx** starts using its internal routing table, and works without Gatekeeper. The **MP-1xx** leaves this Fallback mode as soon as it finds again a responsive Gatekeeper from its list of the Gatekeepers.
8. Alternate Gatekeeper, using RAS IRQ message, can retrieve the state of existing calls. The **MP-1xx** Gateway responds via IRR message.

5.4 Working with H.450 Supplementary Services

H.323 MP-1xx FXS and FXO Gateways support the following supplementary services:

- Hold\Retrieve (H.450.4).
- Consultation\Alternate.
- Transfer (H.450.2).
- Call forward (H.450.3).

To activate supplementary services on MP-1xx Gateway you should Enable Hold, Transfer and Forward services either from Web browser or in *ini* file. To use these features all participated, in a call, H.323 endpoints should comply to the H.450.1-4 standards.

5.4.1 Call Hold and Retrieve Features

- The active call can be put into Hold state by pressing phone's hook-flash button.
- The party that initiates the hold is called the holding party, the other party is called the held party.
- After a successful hold, the held party should hear HELD_TONE and the holding party should hear DIAL_TONE.
- Retrieve can be performed only by the holding party while the call is held and active.
- The holding party performs the retrieve by pressing the hook flash.
- After a successful retrieve the voice should be connected again.

5.4.2 Consultation\Alternate Feature

- The consolation feature is relevant only for the holding party (applicable only for MP-1xx FXS Gateway).
- After holding a call (by pressing hook flash) the holding party hears dial tone and may now initiate a new call which is called a consultation call.
- At the stage of hearing dial tone or dialing to the new destination (before completing the dialing) the user may retrieve the held call by pressing the hook flash.
- The held call can't be retrieved while hearing ring back tone.
- After the consultation call is connected, the user may switch between the held and active call by pressing the hook flash.

5.4.3 Call Transfer

There are two types of call transfers:

- Consultation Transfer
- Blind Transfer

The common way to perform a consultation transfer is as follows:

In the transfer scenario there are three parties:

A - transferring, B – transferred, C – transferred to.

1. A Calls B.
2. B answers.
3. A presses the hook flash and puts B on hold (party B will hear a hold tone)
4. A dials C.
5. After A completed dialing C , he can perform the transfer by on-hook the A phone.

6. After the transfer is completed B and C parties are engaged in a call.

The transfer can be initiated at any of the following stages of the call between A to C:

a. Just after completing dialing C phone number - Transfer from setup.

b. While hearing ring back – Transfer from alert.

c. While speaking to C – Transfer from active.

Blind transfer is performed after we have a call between A and B, and A party decides to transfer the call to C immediately without speaking with C.

The result of the transfer is a call between B and C (just like consultation transfer only skipping the consultation stage).

MP-1xx Gateways can't initiate a blind transfer but they can participate in such transfer as B and C parties.

5.4.4 Call Forward

There are three kinds of call forwards:

1. Immediate – Any incoming call will forwarded immediately and unconditionally.
2. Busy – Incoming call will be forwarded if the endpoint is busy.
3. No reply – The incoming call will be forwarded if it wasn't answered for a specified time.

There are three parties participated in call forward:

1. Served party – the party that is configured to forward the call – MP-1xx FXS.
2. Originating party – the party that initiated the first call – MP-1xx FXS or FXO.
3. Diverted party – the new destination of the forwarded call – MP-1xx FXS or FXO.

The served party (MP-1xx FXS) can be configured from Web browser or *ini* file to activate one of the call forward modes per each of the Gateway's endpoint, including forwarded destination phone number and "no answer" timeout for "No reply" forward.

Figure 5-1: Call Forwarding Table

Gateway Port Number	Forward Type	Forward to Phone Number	Time for No Reply Forward
Port 1	immediate	201	
Port 2	immediate	202	
Port 3	on busy	203	
Port 4	on busy	203	
Port 5	no reply	250	20
Port 6	no reply	251	30
Port 7	not in use		
Port 8	not in use		

5.5 Configuration Examples

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have identical functionality (the **MP-124 and MP-102** Gateways support FXS only), except for the number of channels, and are referred to collectively in this manual as the **MP-1xx**.

Note 2: For simplicity, the **MP-108** 8-port Media Gateway is used in the following examples.

5.5.1 Establishing a Call between Two Gateways

In this demo, using **MP-108 FXS Gateways** as an example, a call can be made from an end-point in one Gateway to an end-point in another Gateway.

➢ To Configure the *ini* file parameters take next 9 steps:

1. Do **NOT** use Gatekeeper.
2. End point numbers of the first Gateway: 101…108.
3. End point numbers of the second Gateway: 201...208.
4. Internal routing prefix table should contain the IP address of the first **MP-108** for numbers starting with 10 and the IP address of the second Gateway for numbers starting with 20
5. Prefix = 10, IP address of first **MP-108**.
6. Prefix = 20, IP address of second **MP-108**.
7. Start **MP-108** Gateways and download the configuration *ini* file
8. Make a call from one end-point to the other.

5.5.2 Using Two Gateways with Gatekeeper

In this demo, a call can be made from an end-point in one Gateway to an end-point in another.

➢ To Configure the *ini* file parameters take next 8 steps:

1. Use Gatekeeper.
2. In “GatekeeperIp”, insert the IP of the PC that runs the Gatekeeper.
3. End point numbers of the first **MP-108** Gateway: 101...108
4. End point numbers of the second **MP-108** Gateway: 201...208
5. Run Gatekeeper on the PC or laptop.
6. Start **MP-108** Gateways and download the configuration ini file
7. Make a call from one end-point to the other.
8. Display the Gatekeeper log messages to show the Gatekeeper activities.
9.

5.5.3 Using Gateway with NetMeeting™

In this demo, a call can be made from an end-point in one Gateway to NetMeeting™.

➢ To Configure *ini* file parameters take next 7 steps:

1. Configure end point numbers of the **MP-108/**FXS Gateway: 101...108
2. Configure NetMeeting IP address using prefix definition:
   Prefix = ’200Netmeeting IP address’.

3. Start **MP-108** Gateway and download the configuration *ini* file
4. Set default number for NetMeeting to **MP-108** calls: DefaultNumber =101 (or any other number 102 to 108).
5. Start **MP-108** Gateway and download the configuration *ini* file
6. Make a call from any **MP-108** end-point to NetMeeting™ by dialing 200
7. Make a call from NetMeeting™ to IP address of **MP-108**. The phone connected to port 1 of the **MP-108** then rings

5.5.4 Remote Extension with FXO & FXS Gateways Example

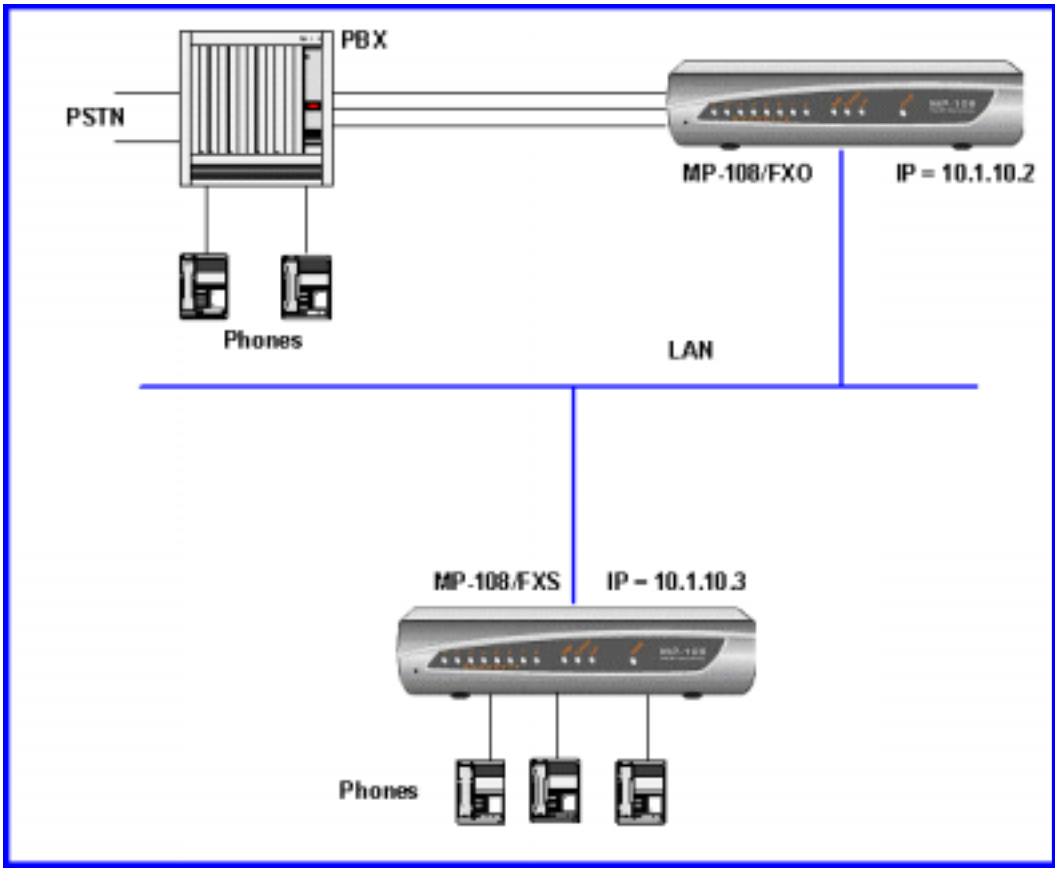
This application explains how to demonstrate remote extension via IP, using **MP-108/FXO** and **MP-108/FXS** Gateways. In this configuration, PBX incoming calls are routed to "Remote Extension" via the **MP-108/FXO** and **MP-108/FXS** Gateways.

➢ Requirements

- One **MP-108/FXO** Gateway
- One **MP-108/FXS** Gateway
- Analog phones (POTS)
- PBX – one or more PBX loop start lines
- LAN.

Connect the **MP-108/FXO** ports directly to PBX lines as shown on the diagram below:

Figure 5-1: MP-108/FXS & MP-108/FXO Layout



5.5.4.1 Dialing from Remote Extension (Phone connected to MP-108/FXS)

➢ **To configure the call take the next 6 steps:**

1. Take the handset off, to hear the dial tone coming from PBX, as if the phone was connected directly to PBX.
2. **MP-108/FXS** and **MP-108/FXO** establish a voice path connection from the phone to the PBX immediately after the phone handset was raised.
3. Dial the destination number (the DTMF digits are sent, over IP, directly to PBX).
4. All tones heard are generated from PBX (such as ringback, busy or fast busy tones).
5. There is one-to-one mapping between **MP-108/FXS** ports and PBX lines.
6. The call is disconnected when the phone connected to **MP-108/FXS** goes on-hook.

5.5.4.2 Dialing from other PBX line, or from PSTN

➢ **To configure the call take the next 5 steps:**

1. Dial the PBX subscriber number the same way as if the user's phone was connected directly to PBX.
2. Immediately as PBX rings into **MP-108**/FXO, the ring signal is "send" to phone connected to **MP-108/FXS**.
3. Once the phone's handset, connected to **MP-108/FXS**, is raised, the **MP-108/FXO** seizes the PBX line and the voice path is established between the phone and the PBX line.
4. There is a one to one mapping between PBX lines and **MP-108/FXS** ports. Each PBX line is routed to the same phone (connected to **MP-108/FXS**).
5. The call is disconnected when phone connected to **MP-108/FXS** goes on-hook.

5.5.4.3 MP-108/FXS Configuration (using the FXS *ini* file)

➢ **To Configure the MP-108/FXS *ini* file take these 4 steps:**

1. Configure in FXS *ini* file the endpoint numbers from 100 to 107.
2. Configure TargetOfChannel table to include phone numbers of **MP-108/FXO** Gateway: such as TargetOfChannel0 = 200. (When phone connected to port #0 goes off-hook, the FXS Gateway automatically dials "200" number).
3. Configure IP to phone table, to IP address and numbers of the FXO Gateway: such as Prefix=20,10.1.10.2 (where 10.1.10.2 is an IP address of **MP-108/FXO**).
4. Set 'IsDialNeeded = 0' to activate automatic dialing, when the handset goes off-hook.

```
IsDialNeeded = 0
;-----------------------------------
; Phone of each end point
;-----------------------------------
ChannelList = 0,8,100

;-----------------------------------
; Automatic dialed numbers
;-----------------------------------
TargetOfChannel0 = 200
TargetOfChannel1 = 201
```

TargetOfChannel2 = 202
TargetOfChannel3 = 203
TargetOfChannel4 = 204
TargetOfChannel5 = 205
TargetOfChannel6 = 206
TargetOfChannel7 = 207

; Phones to IP routing table
;-----------------------------------
Prefix = 20,10.1.10.2

5.5.4.4 MP-108/FXO configuration (using the FXO *ini* file)

➢ **To Configure the MP-108/FXO *ini* file take these 4 steps:**

1. Configure in FXO *ini* file the endpoint numbers from 200 to 207.
2. Configure TargetOfChannel table to include phone numbers of the **MP-108/FXS** Gateway: such as TargetOfChannel0= 100 (when ringing signal is detected at port #0 of FXO Gateway, the FXO Gateway automatically dials "100" number).
3. Configure IP to phone table, to IP address and numbers of the FXS Gateway: such as Prefix=10,10.1.10.3 (where 10.1.10.3 is an IP address of **MP-108/FXS**).
4. Set 'IsDialNeeded = 0' to activate automatic dialing when ringing is detected at FXO port.

IsDialNeeded = 0
;-----------------------------------
; Phone of each end point
;-----------------------------------
ChannelList = 0,8,200
;-----------------------------------
; **Automatic dialed numbers**
;-----------------------------------
TargetOfChannel0 = 100
TargetOfChannel1 = 101
TargetOfChannel2 = 102
TargetOfChannel3 = 103
TargetOfChannel4 = 104
TargetOfChannel5 = 105
TargetOfChannel6 = 106
TargetOfChannel7 = 107

;-----------------------------------
; **Phones to IP routing table**
;-----------------------------------
Prefix = 10,10.1.10.3

Reader's Notes

6 Provisioning

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS Gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO Gateways.

6.1 Provisioning for H.323 Operation

Initial configuration of the **MP-1xx** is provided through Web browser control or by loading of *mp108.ini* configuration file. The configuration *ini* file can be downloaded from Web browser using HTTP or TFTP protocols or by using AudioCodes configuration utility. The *ini* file name is provided in the field 'Boot File Name' of the BootP server.

To create an *ini* file, it is recommended to use the Excel™ utility provided. To use the Excel™ utility, first install the Microsoft™ Office 2000™ Excel™ application.

The *ini* file contains the following information:

- Basic and Logging Parameters shown in Table 6-1 on page 54
- Channel Parameters shown in Table 6-2 on page 56.
- H.323 parameters shown in Table 6-3 on page 59).
- Names for optional Call Progress Tone file. For detailed information, refer to Section 6.4.
- Name for optional Telephony Interface (Coeff.dat) Configuration file. For **MP-1xx/FXS** and for **MP-10x/FXO** two different files should be used. Refer to Section 6.5 for more details.

Note: The names of Call Progress and Coeff.dat files in *ini* file must be enclosed in quotation marks ('…').

All *ini* file data is downloaded at startup and stored in non-volatile memory. The provisioning procedure should be used again only to modify **MP-1xx** parameters; otherwise, BootP and TFTP is not needed again.

The Default Channel Parameters are applied to all **MP-1xx** channels.

Users do not have to specify all parameters, as each unspecified parameter is set to its default value. Using the *ini* file resets all unspecified parameters to their default values.

The Channel Parameters define the DTMF, Fax and Modem transfer modes. Refer to for a detailed description of these modes.

6.1.1 Basic, Logging and Web Parameters

Table 6-1: Basic and Logging Parameters (continues on pages 54 to 55)

ini File Field Name GUI Parameter Name*	Valid Range and Description
MGControlProtocolType	4 = for H.323 Gateway
DSPVersionTemplateNumber	0 = Firmware DSP version supports PCM/ADPCM, G723 and G729 Coders (default) 1 = Firmware DSP version supports PCM/ADPCM, and NetCoder coders 2 = Same as "0" but with voice and energy detectors 3 = Same as "1" but with voice and energy detectors DSP templates 2 or 3 should be selected for FXO Gateway Disconnect supervision feature, enabled by: "EnableSilenceDisconnect = 1".
EthernetPhyConfiguration	0 = 10 Base-T half-duplex 1 = 10 Base-T full-duplex 2 = 100 Base-T half-duplex 3 = 100 Base-T full-duplex 4 = auto-negotiate (Default) Auto-negotiate falls back to half-duplex mode (HD) when the opposite port is not in auto-negotiate, but the speed (10 Base-T, 100 Base -T) in this mode is always configured correctly.
DNSPriServerIP (DNS Primary Server IP)	IP address of primary DNS server
DNSSecServerIP (DNS Secondary Server IP)	IP address of secondary DNS server
DHCPEnable (Enable DHCP)	0 – Disable (default) 1 – Enable After the Gateway is powered up it will try first to communicate with BootP server. If BootP server is not responding and " DHCPEnable =1" the Gateway will send DHCP request to configure its IP address and other network parameters from enterprise DHCP server. The DHCPEnable is a special "Hidden" parameter. Once defined and saved in the flash memory, it is used even if it doesn't appear in the *ini* file.
BootPRetries	1 = Single BootP request. 2 = 2 BootP retries - (3 seconds). 3 = 3 BootP retries - (default, 6 seconds) 4 = 10 BootP retries - (30 seconds). 5 = 20 BootP retries - (60 seconds). 6 = 40 BootP retries - (120 seconds). 7 = 100 BootP retries - (300 seconds). 15 = BootP retries forever. Number of BootP retries, and then DHCP retries (if DHCPEnable = 1) at Gateway startup. The DHCPEnable is a special "Hidden" parameter. Once defined and saved in the flash memory, it is used even if it doesn't appear in the *ini* file.
Note that BootPRetries parameter becomes active after the **MP-1xx** is reset and *ini* file is loaded. To change the parameters, first modify the *ini* file, and then reset the Gateway.	
EnableDiagnostics	0 = No diagnostics (default) 1 = Perform diagnostics
WatchDogStatus	0 = Disable Gateway's watch dog 1 = Enable Gateway's watch dog (default)
EnableLanWatchDog	0 – Disable LAN WatchDog (default) 1 – Enable LAN WatchDog MP-1xx will be restarted if LAN failure is detected
SysLogServerIP (SysLog Server IP Address)	IP address in dotted format notation, e.g., '192.10.1.255'

Table 6-1: Basic and Logging Parameters (continues on pages 54 to 55)

ini File Field Name GUI Parameter Name*	Valid Range and Description
EnableSyslog (Enable SysLog)	0 = Disable SysLog (default) 1 = Enable SysLog If SysLog is disabled all Logs and error messages are sent to RS-232 serial port if "DisableRS232 = 0"
DisableRS232	0 – RS-232 serial port is enabled (default) 1 – RS-232 serial port is disabled To enable sending of all log and error messages to the RS-232 serial port, define: "EnableSyslog = 0" and "DisableRS232 = 0"
LoggerFormat	0 – name + msg 1 – time + msg 2 – name + time + msg 3 – SysLog prefix + msg (default)
DisableWebTask	0 – Enable Web management (default) 1 – Disable Web management
ResetWebPassword	Allows resetting to default of Web password to: Username: "Admin" Password: "Admin"
Disable WebConfig	0 = Enable changing parameters from Web (default) 1 = Operate Web server in "read only" mode
SNMPManagerIP (SNMP Manager IP)	IP address of SNMP Manager. The SNMP manager is used for receiving SNMP Traps. For example: SNMPManagerIP = 10.2.1.10
DisableSNMP (Enable SNMP)	0 – SNMP is enabled (default) 1 – SNMP is disabled
HTTPport	HTTP port used for Web management (default = 80)

* The parameter in the brackets refers to the Embedded Web Server. Some parameters don't appear in the Web.

6.1.2 Channel Parameters

Table 6-2: Channel Parameters (continues on pages 56 to 58)

ini File Field Name GUI Parameter Name*	Valid Range and Description
DJBufMinDelay (Dynamic Jitter Buffer Minimum Delay)	0 to 150 msec (default = 70) Dynamic Jitter Buffer Minimum Delay. (Described in the AudioCodes "VoPLib Reference Library User Manual", Document # LTRT-00744, Section 2.3.18)
DJBufOptFactor (Dynamic Jitter Buffer Optimization Factor)	0 to 12 (default = 7) Dynamic jitter buffer frame error/delay optimization. This is described in the AudioCodes "VoPLib Reference Library User Manual", Document # LTRT-00744 Section 2.3.18
BaseUDPPort (RTP Base UDP Port)	Range 1000-64000 (default 4000) Lower boundary of UDP ports to be used by the Gateway for RTP, RTCP and T.38 channels. The upper boundary is the BaseUDPPort + 10*(number of Gateway's channels). For details refer to the Appendix A
ECHybridLoss	0 = 6 dB (default) 1 = 9 dB 2 = 0 dB 3 = 3 dB Sets the four wire to two wire worst case Hybrid loss, the ratio between the signal level sent to the hybrid and the echo level returning from the hybrid.
FaxModemBypassM (Fax Modem Bypass Packing Factor)	1, 2 (default = 1) Number of 20 msec payloads to be used for generating one RTP fax/modem bypass packet.
FaxModemRelayVolume	-18 to -3, corresponding to -18 dBm to -3 dBm in 1 dB steps. (Default = -12 dBm) Fax gain control.
FaxRelayECMEnable (Fax Relay ECM Enabled)	0 = Disable using ECM mode during Fax Relay 1 = Enable using ECM mode during Fax Relay. (default)
FaxRelayEnhanced RedundancyDepth (Fax Relay Enhanced Redundancy Depth)	0 to 4 (default =2) Number of repetitions applied to control packets when using T.38 standard.
FaxRelayRedundancyDepth (Fax Relay Redundancy Depth)	0 to 2 (default =0) Number of repetitions to be applied to each fax relay payload when transmitting to network (applicable only when T38ProtectionMode = 0).
FaxRelayMaxRate (Fax Relay Max Rate (bps))	Limits the maximum rate at which fax messages are transmitted. 0 = 2.4 kbps 1 = 4.8 kbps 2 = 7.2 kbps 3 = 9.6 kbps 4 = 12.0 kbps 5 = 14.4 kbps, (default).
FaxTransportMode (Fax transport Mode)	Sets the Fax transport 0 = disable 1 = relay, (default) 2 = bypass.
UseT38orFRF11	0 = Use proprietary FRF.11 syntax to send/receive fax relay. 1 = Use T.38 protocol to send/receive fax relay, (default).
V21ModemTransportType (V21 Transport Type)	0 = Transparent, (default) 2 = ModemBypass.
V22ModemTransportType (V22 Transport Type)	0 = Transparent 2 = ModemBypass, (default).
V23ModemTransportType (V23 Transport Type)	0 = Transparent 2 = ModemBypass, (default).
V32ModemTransportType **(For V.32 & V.32bis modems)** (V32 Transport Type)	0 = Transparent 2 = ModemBypass, (default).

Table 6-2: Channel Parameters (continues on pages 56 to 58)

ini File Field Name GUI Parameter Name*	Valid Range and Description
V34ModemTransportType **(For V.34 & V.90 modems)** (V34 Transport Type)	0 = Transparent 2 = ModemBypass, (default).
FaxModemBypassCoderType (Fax/Modem Bypass Coder Type)	Coder to be used while performing fax/modem bypass. Refer to acTCoders enumeration. Usually, high bit rate coders such as G.711 and G.726/G.727 should be used.
	0 = G711 A-law =0, (default)
	1 = G711 μ-law=1,
	4 = G726_32
	11 = G727_32.
FaxBypassPayloadType (Fax Bypass Payload Type)	Fax Bypass RTP dynamic payload type Default = 102
T38ProtectionMode	0 = Use redundancy packets for protecting T.38 fax relay stream, (default) 1 = Use Forward Error Correction (FEC) algorithm to protect T.38 fax relay stream (isn't implemented)
DTMFVolume (DTMF Volume)	-31 to 0, corresponding to -31 dBm to 0 dBm in 1 dB steps (default = -11 dBm) DTMF gain control.
DTMFTransportType (DTMF Transport Type)	0 = erase digit from voice stream, do not relayed to remote. 1 = erase digit from voice stream, relay to remote. (Default) 2 = digits remains in voice stream. 3 = erase digit from voice stream, relay to remote according to RFC 2833 standard 7 = digits will be send using RFC 2833 standard, but received RFC 2833 digits will be muted from audio.
MFTransportType (MF Transport Type)	0 = erase MFs from voice transport, not relayed to remote. 1 = erase MFs from voice transport, relay to remote. (Default) 2 = MFs are not erased from voice, not relayed to remote.
RFC2833PayloadType (RFC 2833 Payload Type)	The RFC 2833 DTMF relay dynamic payload type. Range: 96 to 99, 106 to 127; Default = 96 The 100, 102 to 105 range is allocated for AudioCodes proprietary usage. Cisco™ are using payload type 101 for RFC 2833. The same payload type should be used for receive and for transmit.
InputGain (Input Gain)	-31 to 31 corresponding to -31 dB to +31 dB in 1 dB steps. (Default = 1 dB). PCM input gain.
RTPRedundancyDepth (RTP Redundancy Depth)	0 = Disable redundancy packets generation (default) 1 = Enable generation of RFC 2198 redundancy packets.
VoiceVolume (Voice Volume)	-31 to 31, corresponding to -31 dB to +31 dB in 1 dB steps. (Default = 1 dB). Voice gain control
M (Packing Factor)	Number of codec payloads (5,10, 20 or 30 msec, depending on selected codec) to be used for generating one RTP packet. M = n payloads (n = 1, 2 or 3); M = 1 (default)
SCE (Silence Suppression)	0 = silence compression disabled (default) 1 = silence compression enabled
ECE (Echo Canceller)	0 = Echo Canceler disabled 1 = Echo Canceler Enabled (default)
IPDiffServ (RTP IP Diff. Serv)	0 – 63 value for setting the Diff Services Code Point (DSCP). If defined it will override the IP TOS and IP Precedence settings. Applies only to RTP packets
IPPrecedence (RTP IP Precedence)	0 to 7 (default 0) Sets the value of the IP precedence field in the IP header for all RTP packets
IPTOS (RTP IP TOS)	0 to 15 (default 0) Sets the value of the IP Type Of Service field in the IP header for all RTP packets
FlashHookPeriod	300 to 1500 (default 700) Flash Hook time in msec. The parameter is used for Flash Hook detection in **MP-1xx**/FXS and for Flash Hook generation in **MP-1xx**/FXO Gateways.

Table 6-2: Channel Parameters (continues on pages 56 to 58)

ini File Field Name GUI Parameter Name*	Valid Range and Description
MinimumFlashHookTime	25-300, (default = 250) Minimum threshold in msec + 50msec for detection of HookFlash. Relevant only for MP-1xx FXS Gateways.
DTMFDetectionPoint	0 = DTMF event is reported when button is pressed 1 = DTMF event is reported on button release (default) The parameter is used for out of band dialing (H.245 user input message or H.225 keypad facility)
DTMFDigitLength	Time in msec for generating DTMF to PSTN side. Default = 100 msec
DTMFInterDigitInterval	Time in msec between generated DTMFs to PSTN side Default = 100 msec
CallerIDType	0 = Bellcore GR-30-CORE Type 1(default) 1 = ETSI Type 1 Both Bellcore and ETSI Type 1 Caller ID signals are generated/detected between first and second rings.
TestMode	0 = CoderLoopback, encoder-decoder loopback inside DSP. 1 = PCMLoopback, loopback the incoming PCM to the outgoing PCM. 2 = ToneInjection, generates a 1000 Hz tone to outgoing PCM. 3 = NoLoopback, (default).

* The parameter in the brackets refers to the Embedded Web Server. Some parameters don't appear in the Web.

6.1.3 H.323 Parameters

Table 6-3: H.323 Parameters (continues on pages 59 to 67)

ini File Field Name GUI Parameter Name*	Valid Range and Description
GatewayVersion (Gateway ID)	Version of Gateway, for example "GatewayVersion = 4.2 Beta".
StaticNatIP (NAT IP Address)	Static NAT IP address. Global Gateway IP address. Should be defined if static Network Address Translation device (NAT) is used between the Gateway and Internet.
IsGatekeeperUsed (Working with Gatekeeper)	0 = no Gatekeeper used (internal phones table used) 1 = Gatekeeper used (default)
GatekeeperIp (Gatekeeper IP Address)	IP address of Gatekeeper Used if IsGatekeeperUsed = 1 GatekeeperIP parameter can contain up to three IP addresses. In next example there are 2 redundant gatekeepers in addition the first gatekeeper. GatekeeperIP = 10.2.37.10 GatekeeperIP = 10.2.37.11 GatekeeperIP = 10.2.37.12 To enable the use of redundant gatekeeper option, select "IsRedundantGKUsed = 1".
GatekeeperID (gatekeeper ID)	Gatekeeper Identification name. Used in Registration Request (RRQ messages). (Default = not configured).
IsRedundantGKUsed (Gatekeeper Redundancy)	0 = No redundancy 1 = Allow gatekeeper redundancy, according to GatekeeperIP addresses. If redundancy is used, Gateway tries to communicate with redundant Gatekeepers, if no response from the current Gatekeeper. If a new Gatekeeper is found the Gateway stays working with it, until the next failure.
GWRegistrType (Gateway Registration Type)	0 – E.164 (default) 1- H.323-ID 2- E.164 and H.323-ID If H323IDString is configured, it is added to GW registration if GWRegistrType=1 or 2.
RegistrationPrefix (Registration Prefixes table)	The registration prefix is used for **MP-10x**/FXO Gateway registration to gatekeeper. It can appear up to 5 times. For example: RegistrationPrefix = 10
H323IDString (H.323 ID)	Definition of Gateway H.323-ID for registration to GK. In case that H323IDString exist in configuration and IsGatekeeperUsed=1, GW will send registration to GK with this H323-ID string. Up to 19 characters per string.
SourceEncodeMode (Source Number Encoding Type)	0 – E.164 (default) 1- H.323-ID 2- E.164 and H.323-ID
DestEncodeMode (Destination Number Encoding Type)	0 – E.164 (default) 1- H.323-ID 2- E.164 and H.323-ID
RegistrationTime (registration Time [sec])	Time in seconds between registrations to gatekeeper (default 60)
ResponseTimeOut (RAS Response Timeout [sec])	Time in seconds, the Gateway waits for RAS response from GK, before message retransmission (default 2).
MaxRetries (Number of RAS Retransmissions)	Number of retransmissions of RAS message to GK, before Gateway decides that the GK is not responding (default 2). If "IsRedundantGKUsed = 1", the Gateway tries to use redundant gatekeeper from its list.
TimeBetweenGKsLoops (Time between GK Retries)	Time in seconds before the Gateway tries again to go through the gatekeeper list (default 60).
IsGKFallbackUsed (Fallback to Internal Routing)	0 = Fallback is not used (default) 1 = If gatekeeper(s) doesn't respond, use internal Phone to IP table for call routing. The Gateway leaves the fallback mode as soon as it again finds the gatekeeper.

Table 6-3: H.323 Parameters (continues on pages 59 to 67)

ini File Field Name GUI Parameter Name*	Valid Range and Description
CanMapAliases (Enable Mapping of Alias Numbers)	If enabled, GK can change Gateway destination number, using Alias parameter in ACF message.
IsAutoDiscovery	0 = no auto discovery of Gatekeeper (default) Gatekeeper Auto Discovery requires multicast and is not supported
IsTerminal	0 = Standard Gateway behavior (default) 1 = Gateway imitates an H.323 terminal with multiple aliases (up to 24 for **MP-124**). In all Gateway messages the “terminal type” value is set to “terminal”
IsTunnelingUsed	0 = no tunneling (default) 1 = tunneling
EnableRAI	0 – Disable RAI messages 1 – RAI message with almost out of resources will be send to GK if GW resources are below the threshold. Usually RAI will be applicable if Gateway is configured to use Hunt groups.
RAIHighThreshold	0-100, "Out of resources" Gateway threshold, in percentage [default = 90%] When number of GW busy endpoints [%} is above the High Threshold, GW will send RAI message with "almostOutOfResources = TRUE " GW Resources [%] is calculated as number of busy endpoints divided by total available GW endpoinrs.
RAILowThreshold	0-100, "Out of resources" Gateway threshold, in percentage [default = 90%] When number of GW busy endpoints [%] is below the LowThreshold, GW will send RAI message with "almostOutOfResources = FALSE "
RAILoopTime	GW check for resource availability each RAILoopTime period [sec] (default 10sec)
IsFastConnectUsed	0 = normal connect (default) 1 – fast connect
CoderName (Coders)	Coder name that is used. g711Ulaw64k – G.711 μ-law, 20 msec g711Alaw64k – G.711 A-law, 20 msec g711Ulaw64k,5 – G.711 μ-law, 5 msec g711Alaw64k,5 – G.711 A-law, 5 msec g711Ulaw64k,10 – G.711 μ-law, 10 msec g711Alaw64k,10 – G.711 A-law, 10 msec g7231 – G.723 6.3 kbps (default) g7231r53 – G.723 5.3 kbps g726 – G.726 ADPCM 16 kbps (Payload Type = 35) g726r16 – G.726 ADPCM 16 kbps, Cisco mode (PT=23) g726r32 – G.726 ADPCM 32 kbps (PT=2) g729 – G.729A , 20msec g729,10 – G.729, 10msec NetCoder6_4 – NetCoder 6.4 kbps NetCoder7_2 – NetCoder 7.2 kbps NetCoder8 – NetCoder 8.0 kbps NetCoder8_8 – NetCoder 8.8 kbps This parameter can appear up to 5 times.
ChannelList	List of phone numbers for **MP-1xx** channels a, b, c a = first channel b = number of channels starting from “a” c = the phone number of the first channel example: ChannelList = 0,8,101 Defines phone numbers 101 to 108 for up to 8 **MP-108** channels. The *ini* file can include up to ten “ChannelList = “ entries The “ChannelList = “ can be used instead or in addition to Channel2Phone parameter.

Table 6-3: H.323 Parameters (continues on pages 59 to 67)

ini File Field Name GUI Parameter Name*	Valid Range and Description
Channel2Phone	Phone number of channel. Its format: Channel2Phone = "<channel>, <number>" <channel> is 0...23. Example: "Channel2Phone = 0, 1002" Appears once for each channel: 8 times for **MP-108**, or 4 times for **MP-104** and twice for **MP-102**. For 8-port and 24-port Gateways it is suggested to use "ChannelList = " parameter, where in a single line, all Gateway's phone numbers can be defined. The "Channel2Phone" can be used instead or in addition to "ChannelList = " parameter.
TrunkGroup_x (Trunk Group Table)	a-b,c or just a,c x – Hunt group ID, starting with 1 a – Starting port (from 1) b – Ending port (up to 24 for MP-124) c – phone number allocated for the first port For example: TrunkGroup_1 = 1-4,100 TrunkGroup_2 = 5-8,200 Trunk Group can be used instead of ChannelList and Channel2Phone parameters. It can be used for both FXS and FXO Gateways.
EnableHold (Enable Hold)	0 – Disable Hold 1 – Enable Hold service (H.450.4) If enabled the Hold (or unhold) is activated at FXS Gateway by HookFlash. On receipt of Hold request, the remote party is put on hold, and plays hold tone to tel port.
EnableTransfer (Enable Transfer)	0 – Disable Call Transfer (default) 1 – Enable call transfer service (H.450.2) Transfer is initiated at FXS Gateway by HookFlash signaling. If service is enabled the remote party will perform the call transfer.
EnableForward (Enable Forward)	0 – Disable call forward (default) 1 – Enable call forward service (H.450.3). For MP-1xx FXS Gateways a "Call forward table" should be used to define call forward modes (refer to the ForwardInfoX parameter).
ForwardInfoX	a,b,c x – Gateway port number (1 to 24) a – Call forward mode 0 – not in use 1 – On busy 2 – Immediate forward 3 - No reply for time "c" b – Forwarded number c – Timeout [sec] for no reply Incoming IP→Tel call will be forwarded to "b" number on one of the conditions: busy, immediate or no reply. Applicable only for MP-1xx/FXS Gateways
IsFaxUsed	0 = No 1 = Fax is send/received using H.323 Annex D T.38 procedure.
DetFaxOnAnswerTone	0 = Receiving Gateway initiates T.38 on receiving HDLC preamble signal from Fax (default) 1 = Receiving GW initiates T.38 on receiving of CED answer tone from Fax.

Table 6-3: H.323 Parameters (continues on pages 59 to 67)

ini File Field Name GUI Parameter Name*	Valid Range and Description
ChannelToFax#	Type of H.323 (annex D) FAX channel 0 = Voice only 1 = Fax only (can be used only with Fast Start or Tunneling) 2 = Voice + Fax (default) Example: ChannelToFax7 = 1 (use port '7' of **MP-108** only for Fax) 'IsFaxUsed' parameter should be set to '1' in order to use the H.323 Fax according to Annex D T.38. When channel is set to 2 (Voice + Fax), at beginning of a call, the logical channel is opened as Voice channel, and after Fax detection, the channel is reopened as Fax T.38 channel. This parameter can appear up to 8 times.
Prefix	Mapping phone number to IP address, using phone number prefix Example: Prefix = 20,10.2.10.2 Any dialed number that starts with "20" is routed to IP address "10.2.10.2". Needed when Gatekeeper is not used. Can appear up to 20 times. Maximal prefix size is 7 digits
PSTNPrefix (IP to Trunk group routing table)	a,b a – destination number prefix (for IP→Tel calls) b – Hunt group ID Outgoing IP→Tel calls starting with "a" will be routed to Hunt group "b"
IsDialNeeded (Enable Automatic Dialing)	0 = no dial needed (automatic dialing) 1 = dial needed (default) If 0 = TargetOfChannel parameters define the automatic dialed number. This parameter is applicable for both FXS and FXO Gateways. If "DialisNeeded =1" the FXO Gateway will seize the line (after detecting the ringing signal), play a dial tone, collect DTMF digits and send Setup to IP destination.
TargetOfChannel# (Automatic Dialing Table)	Automatic dialed phone number. The automatic dialed number, used if OFF HOOK detected in FXS channel, or ringing signal is detected in FXO channel. Applicable when IsDialNeeded = 0. Its format: "TargetOfChannel<channel> = <number>". Example: "TargetOfChannel1 = 123" The parameter, if used, should be defined per Gateway FXS or FXO port (channel)
IsUseFreeChannel (Selects Next Available Channel)	0 = Select the FXO channel according to destination phone number received in H.225 setup message, (default) 1 = Select the next available FXO channel Used for IP→ **MP-1xx**/FXO calls The next available FXO channel is selected, out of the Gateway channels defined in 'ChannelList' and Channel2Phone' parameters. When using one stage dialing, ('IsTwoStageDial =0'), 'IsUseFreeChannel' should be equal to '1'. For one stage dialing the **MP-1xx**/FXO selects the next free channel, and dials into the FXO line the destination phone number received in Setup message.
IsTwoStageDial	0 = One stage dialing 1 = Two Stage Dialing (default) Used for IP → **MP-10x**/FXO calls For 'Two Stage Dialing the **MP-10x**/FXO seizes the PSTN/PBX line, without performing any dial, the remote User is connected over IP to PSTN/PBX, and all further signaling (dialing and call progress tones) is done directly with the PBX without Gateway intervention. For 'One Stage Dialing' **MP-10x**/FXO seizes the next available channel ('IsUseFreeChannel' should be '1'), and dials the destination phone number received in Setup message. Use the 'IsWaitForDialTone' parameter to specify whether the dialing should come after detection of dial tone, or immediately after seizing of the line.

Table 6-3: H.323 Parameters (continues on pages 59 to 67)

ini File Field Name GUI Parameter Name*	Valid Range and Description
IsWaitForDialTone	0 = don't wait for dial tone 1 = Wait for dial tone (default) Used for **MP-1xx**/FXO, for 'One Stage Dialing'. If IsWaitForDialTone = 0, **MP-10x**/FXO dials phone number immediately after seizing the PSTN/PBX line, without 'listening' to dial tone. If IsWaitForDialTone = 1, **MP-10x**/FXO dials phone number only after it detects a dial tone (it can take 3-5 sec to detect a dial tone). The correct dial tone parameters should be configured in call progress tone file.
FXOWaitForDialTime	FXO dialing timeout [in msec] , after seizing the line and before start dialing. (Default =1000) Applicable for MP-10x FXO for single stage dialing.
EnableSilenceDisconnect	If enabled, the FXO Gateway will disconnect call if silence is detected for 120 seconds (default)
FarEndDisconnectPeriod	Duration of Silence period [in seconds] for call disconnection. (Default = 120) Applicable for FXO Gateway, that are using DSP template = 3 or 4.
MaxDigits (Max Digits in Phone number)	2 to 19 (default 4). Maximum number of digits that can be dialed. Dialing ends when maximum number of digits dialed or timeout between digits expired (TimeBetweenDigits parameter), or '#' is dialed.
TimeBetweenDigits (Interdigit Timeout [sec])	0 to 5 (default 4) Inter-digit timeout in seconds, used to terminate dialed numbers.
IsSETUPIncludeNum (Does Setup Include Phone Number)	0 = Send empty SETUP (or GK ARQ) message, without called party number. 1 – Don't send empty setup message (default)
DefaultNumber (Default Destination Number)	Phone number Gateway Dial to this number if SETUP received with no phone number. Must be one of the phone numbers of the end points.
EnableQ931Cause	0 – H.225 Release reason will be send in Release Complete H.323 message 1 – Q.931 cause will be send in Release Complete message (default)
IsSetupAckUsed (Release Call on "Setup ACK")	0 = On receiving of Setup Ack message, Gateway releases the Call (default) 1 = On receiving of Setup Ack message the call is not released. When working in non-overlap mode ('IsOverlapUsed = 0'), use this parameter to enable receiving of Setup Ack messages. This parameter is used for specific, non-standard applications. Usually when working in non-overlap mode Setup Ack messages are not sent.
OpenH245onFS	0 = Don't open H.245 channel of Fast Start (default) 1 = H.245 channel is opened immediately after Fast Start connection is established. Opening of H.245 channel may be needed for relaying DTMF digits over H.245 channel, during a call.
IsDTMFUsed (Enable DTMF over H.245)	0 = not used (default) 1 = DTMF digits are sent over H.245 channel using Facility message Incoming facility messages of out-of-band DTMF is played into FXS or FXO channels. For using out of band DTMF the User must disable inband DTMF by setting: 'DTMFTransportType = 0' (erase digit and don't relay). Note that this feature is not supported when using Fast Start, unless other party opens H.245 channel.
IsHookFlashUsed (Enable Hook-Flash)	0 = not used (default) 1 = Send H.245 User Input Indication message, when hookflash is detected Received facility message with HookFlash signal generates HookFlash at FXO channel
Is931MsgUsed (Send DTMF over H.225/Q.931)	0 = not used (default) 1 = DTMF digits and HookFlash signal is sent in H.225/Q.931 info message (keypad facility) Note that incoming Q.931 DTMF digits and HookFlash are always ignored. To use H.225/Q.931 messages, set the following params: 'IsDTMFUsed = 0' and 'IsOverlapUsed = 0'

Table 6-3: H.323 Parameters (continues on pages 59 to 67)

ini File Field Name GUI Parameter Name*	Valid Range and Description
IsSpecialDigits	0 = not used (Default) 1 = Allows user to dial "*" and "#", and allows endpoint telephone number to contain such digits. Notes: a. When this feature is activated, "#" can't be used to end the dialed number. b. # and * can be used as first digit of dialed number, disregard the value of "IsSpecialDigits" parameter.
IsFSOpenMediaOn Connect	0 = Voice channel is open after sending an Alert message (default) 1 = Voice channel is opened after sending a Connect message After receiving Setup message the Gateway can open voice channel immediately, or it can wait until the call is answered and Connect message is sent. Note that this is only relevant if Fast Start or Tunneling are used.
IsFSMediaInfoSendOn Connect	0 = Fast Start Structure response is send in Alert message (default) 1 = Fast Start Structure response is send in Connect message After receiving Fast Start Setup message, Gateway should reply with H.225 message that includes media information structure. It can be send with Alert or Connect messages. Sending this information in Alert message enables the remote side to open the voice channel before receiving the Connect message. Note that this is only relevant if Fast Start is used.
IsFSRemoteRBTone	0 = Originator of Fast Start call plays local ringback tone, until it receives Connect message (default) 1 = If the received Alert message contains media information structure, the Gateway opens voice channel, without playing any local tone to the channel. It is the remote Gateway's responsibility to play the ringback tone over IP. If the Alert message doesn't contain media information structure, the voice channel is not opened and a local ringback tone is generated, until Connect message is received. Note that this is only relevant if Fast Start is used.
PlayRBTone2Tel	0 – Don't play for fast start 1 – Play 2 – Play according to Progress Indicator (PI). The Gateway will not play local ringback tone if PI=1 or PI=8 was received in H.225 Alert message. (default)
PlayRBTone2IP	0 – Don't play Ringback to to IP (default) 1- Play Ringback tone to IP, and set PI=8 in H,225 Alert message
ProgressIndicator2IP	0 – No PI will be send in H.225 messages 1- PI = 1 will be send to IP in H.225 Alert 8 - PI = 8 will be send to IP in H.225 Alert If not configured at all (default) the PI send in H.225 messages will be set according to "Play Ringback tone to IP" parameter.
CallerDisplayInfo# (Caller ID table)	Caller DisplayInfo table is used to send Caller Identification information per FXS Gateway port to remote IP terminal. This parameter can appear up to eight times (for **MP-10x**), with up to 18 characters per string. For **MP-124** the parameter can appear up to 24 times, with up to 10 characters per string. For example: CallerDisplayInfo0 = AudioCodes 1001 CallerDisplayInfo1 = AudioCodes 1002 ; ; CallerDisplayInfo8 = AudioCodes 1008
EnableCallerID (Enable Caller ID)	0 = Don't send CallerID signal to Gateway's FXS port (default) 1 = Calling number and Display text is sent to Gateway FXS port, between first and second rings, to be displayed on phone's caller ID display for incoming call. In FXO Gateway if "EnableCallerID=1", the Caller ID signal will be detected and send to IP in H.323 Setup message (as "Display" element).
H225ListenPort (H.225 Listen Port)	H.225 TCP listen port (default 1720) Gateway expects to receive H.225/Setup message on this port
H225DialPort (H.225 Dial Port)	H.225 TCP destination port (default 1720) Gateway sends H.225/Setup message to this port

Table 6-3: H.323 Parameters (continues on pages 59 to 67)

ini File Field Name GUI Parameter Name*	Valid Range and Description
H245InitTimeOut	Timeout in seconds for establishment of H.245 connection (Default 30 sec).
RTPMultiplierUsed	0 = The Coder Packetization period in Terminal Capability Set is set to basic coder period, such as 30 msec for G.723. 1 = The Coder Packetization period in Terminal Capability Set takes into consideration the Multiplication Factor [M]. For example if M=2, packetization period of G.729 coder is set to 40 msec (default).
EnableReversalPolarity (Polarity Reversal)	0 = The line polarity is not changed on answer (default) 1 = The FXS GW will change the line polarity on call answer and change it back on call release. The FXO GW will send H.225 CONNECT message on detection of Polarity reversal signal.
EnableCurrentDisconnect (Enable Current Disconnect)	0 –No current disconnect (default) 1 - FXS Gateway will generate a "Current disconnect pulse" of 900msec after a call is released from IP. Applicable only for MP-1xx FXO Gateways.
TimeForReorderTone	Duration of played reorder tone in seconds (default 5 sec). Applicable for FXO port. The tone is played before releasing the FXO line.
TimeForDialTone (Dial Tone Duration)	Duration of played dial tone (default 16 sec), The dial tone is played at FXS Gateway port, after phone is picked up, or after the FXO Gateway seizes the line in respond to ringing. During play of the dial tone, Gateway waits for DTMF digits. Applicable for both FXS and FXO Gateways when "Automatic dialing" feature is disabled, "IsDialNeeded = 0"
EnableQ931Multiplexing (Q.931 Multiplexing)	0 = H.323 Q.931 multiplexing is disabled (default) 1 = H.323 Q.931 multiplexing is enabled When multiplexing is enabled, the H.323 Gateway uses same socket for all H.225 messages send to same destination.
H323BasePort (H.323 Base Port)	Starting TCP/UDP transport port for H.225/H.245 messages. H.323 Gateway will use dynamic ports (except RTP ports) starting from H323Baseport to H.323BasePort+500. For example if H323BasePort = 10000. H.323 Gateway will use dynamic ports within the range 10000 to 12000. These ports are used for RAS, H.225 and H.245 protocols, not including RTP ports. If not specified, the default port range: 32000 to 65000
SecureCallsFromIP (IP Security)	0 = Accept all H.323 Calls (default) 1 = Accept H.323 calls only from IP addresses defined in **Prefix** routing table. When SecureCallsFromIP is enabled, Gateway will reject calls from unknown IP addresses. When using Gatekeeper in routed mode it is recommended to set this parameter to "0", leaving to Gatekeeper the decision to accept or reject H.323 calls.
DestNumberType	0 = Unknown (default) 1 = International Number 2 = National Number 3 = Network Specific Number 4 = Subscriber Number (or local) 6 = Abbreviated Number 7 = Reserved for extension The Number Type (TON) parameter is used in H.225 Setup messages. Not all combinations of TON/NPI are allowed. Usually, you need to select one of the following TON/NPI sets: 0/0 (Unknown/Unknown), 1/1 (International/ISDN) 2/1 (National/ISDN) 4/1 (Subscriber/ISDN) Note: Numbering Plan/Type in the Number Manipulation table, if presented, will override the value defined by the global parameter. Using the Number Manipulation tables enables configuration of the Numbering Plan and Type on a per call basis, according to destination (or source) number.
DestNumberPlan	0 = Unknown (default) 1 = ISDN/Telephony Numbering Plan

Table 6-3: H.323 Parameters (continues on pages 59 to 67)

ini File Field Name GUI Parameter Name*	Valid Range and Description
	3 = Data Numbering Plan 4 = Telex Numbering Plan 8 = National Standard Numbering Plan 9 = Private Numbering Plan 15 = Reserved for extension The Number Plan (NPI) parameter is used in H.225 Setup messages. Not all the Combinations of TON/NPI are allowed. Usually, you need to select one of the following TON/NPI sets: 0/0 (Unknown/Unknown), 1/1 (International/ISDN) 2/1 (National/ISDN) 4/1 (Subscriber/ISDN) Note: Numbering Plan/Type in the Number Manipulation table, if presented, will override the value defined by the global parameter, Using the Number Manipulation table enables configuration of the Numbering plan and Type on a per call basis, according to destination (or source) number.
SourceNumberType	Same as DestNumberType
SourceNumberPlan	Same as DestNumberPlan.
NumberMapTel2IP (Manipulation Tables: TEL → IP Destination Numbers)	a,b,c,d,e,f a = Phone number prefix b = Number of stripped digits c = New prefix to be added, or "none" d = Number of remained digits from right e = H.225/Q.931 Number Plan f = H.225/Q.931 Number Type The parameter defines rule(s) for changing the destination phone number, received in Tel→ IP incoming call. Each rule is applied to a number that matches the "a" prefix. The "b" and "c" actions apply to a number that starts with "a" prefix. The actions apply to the number in the following order: "b", "d" and then "c". Parameters can be skipped by "$$", such as NumberMapTel2IP=01,2,972,$$,0,0
NumberMapIP2Tel (Manipulation Tables: IP → TEL Destination Numbers)	a,b,c,d a = Phone number prefix b = Number of stripped digits c = New prefix to be added, or "none" d = Number of remained digits from right The parameter defines rule(s) for changing the destination phone number, received in IP→Tel call. Each rule is applied to a number that matches the "a" prefix. The "b" and "c" actions apply to a number that starts with "a" prefix. The actions apply to the number in the following order: "b", "d" and then "c".
SourceNumberMapTel2IP (Manipulation Tables: TEL → IP Source Numbers)	Same rules as for destination number
SourceNumberMapIP2Tel (Manipulation Tables: IP → TEL Source Numbers)	Same rules as for destination number
RouteModeIP2Tel (IP to Tel Routing Mode)	0 – Route calls before number manipulation (default) 1 – Route calls after number manipulation Defines order between routing calls to Hunt group and manipulation of destination number.
RouteModeTel2IP (Tel to IP Routing Mode)	0 – Route calls before number manipulation (default) 1 – Route calls after number manipulation Defines order between routing incoming calls to IP, using routing table, and manipulation of destination number. Not applicable if GK is used.
EnableBusyOut	0 – Not used (default) 1 – If LAN or GK is not responding, the Gateway will play a reorder tone on phone OffHook.

Table 6-3: H.323 Parameters (continues on pages 59 to 67)

ini File Field Name GUI Parameter Name*	Valid Range and Description
AddTrunkGroupAs Prefix	0 – not used 1 - For Tel→IP incoming call, Hunt Group ID will be added as prefix to destination phone number. Applicable only if Hunt group ID are configured. Can be used to define various routing rules.
EnableCDR	0 – CDR is not used (default) 1- Call detail record will be send to Syslog server at each Call end.
MaxActiveCalls	The parameter can be used to limit number of calls on GW. When number of concurrent calls reaches MaxActiveCalls, the GW reject calls from TEL and from IP.

* The parameter in the brackets refers to the Embedded Web Server. Some parameters don't appear in the Web.

6.1.4 Loading Configuration Files

Users can use the *ini* file in order to specify Call Progress Tone table files and Line Characteristics control file to be downloaded to the **MP-1xx** during the configuration phase, either directly from the Web browser or by using TFTP procedure. It is also possible to define whether the downloaded files are stored in non-volatile memory so the TFTP process is not required every time the Gateway boots up.

The following *ini* file fields are related to this operation:

"**CallProgressTonesFilename**"	– The name (and path) of the file containing the call progress tones definition. Refer to Section 6.4.4 for additional information on how to create and download this file.
"FXSCoefFileName"	– The name (and path) of the file providing the FXS line characteristic parameters.
"FXOCoefFileName"	– The name (and path) of the file providing the FXO line characteristic parameters.
"BurnCallProgressTonesFile"	– Stores the call progress tones configuration file in non-volatile memory, if set to 1.
"BurnCoefFile"	– Stores the line characteristics file in non-volatile memory, if set to 1.

6.2 The *ini* File Structure

The *ini* file can contain any number of parameters. The parameters are divided into groups by their functionality. The general form of the *ini* file is shown below.

Figure 6-1: *ini* File Structure

```
[Sub Section Name]

Parameter_Name = Parameter_Value
Parameter_Name = Parameter_Value
.
..

; REMARK

 [Sub Section Name]
      .
```

6.2.1 The *ini* File Structure Rules

- Lines beginning with a semi-colon ';' (as the first character) are ignored.
- Carriage Return must be the final character of each line.
- Number of spaces before and after "=" is not relevant.
- If there is a syntax error in the parameter name, the value is ignored.
- Syntax errors in the parameter value field can cause unexpected errors (because parameters may be set to the wrong values).
- Sub-section names are optional.
- The File name String parameters, should be placed between two inverted commas ('…'). For example CallProgressTonesFileName = 'cpusa.dat'
- The parameter field is NOT case sensitive.
- Parameter values should be entered only in decimal format, except for the Call Agent IP address.
- The *ini* file should be ended with one or more carriage returns.
- "[Files] " line should precede the CallProgressTonesFileName, and these lines, if used, should be placed at the end of *ini* file; (refer to the *ini* file example below in Figure 6-2).

6.2.2 The *ini* File Example

An example of an *ini* file for an H.323 Gateway is shown in Figure 6-2.

Figure 6-2: H.323 *ini* File Example

```
MGControlProtocolType = 4

[Channel Params]
DJBufferMinDelay = 75
RTPRedundancyDepth = 1

IsGatekeeperUsed = 1
GatekeeperIp = 192.168.122.179
DefaultNumber = 101
MaxDigits = 3
CoderName = g7231
IsFastConnectUsed = 1

;Phone of each end point
Channel2Phone = 0, 101
Channel2Phone = 1, 102
Channel2Phone = 2, 103
Channel2Phone = 3, 104

EnableSyslog = 0
LoggerFormat = 1

[Files]

CallProgressTonesFilename = 'CPUSA.dat'
BurnCallProgressTonesFile = 1
FXSCOEFFILENAME = 'coeff.dat'
BurnCoefFile = 1
```

Note 1: Using Windows Properties Display, verify that the MS-DOS name of the *ini* file is in fact *mp108.ini*, and NOT by mistake *mp108.ini.ini,* or mp108~.ini.

Note 2: To restore **MP-1xx** default configuration parameters, use the *mp1xx.ini* file without any valid parameters or with semicolon (;) character preceding all lines in the file.

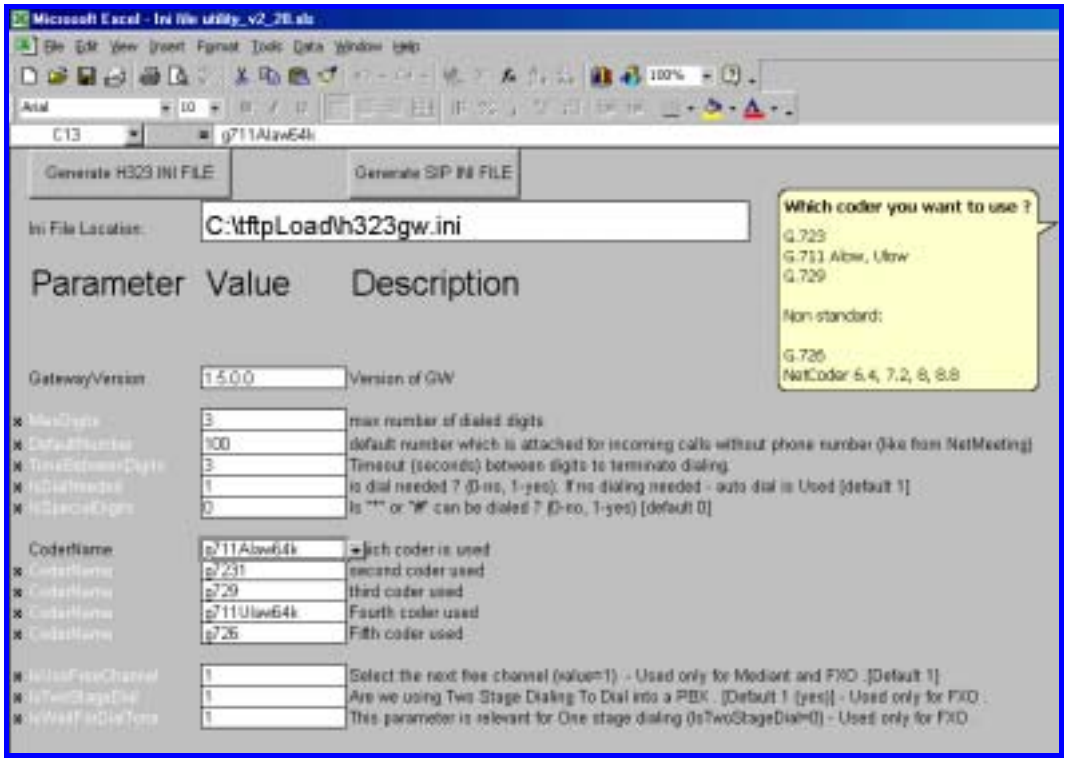
6.3 Excel Utility for *ini* File Generation

The Excel™ Utility enables easy generation of **MP-1xx** and other MediaPack series Gateway *ini* files. To use the Excel utility, first install the Microsoft™ Office 2000 Excel™ application.

Currently the utility can be used to generate *ini* files only for H.323 and SIP Gateways.

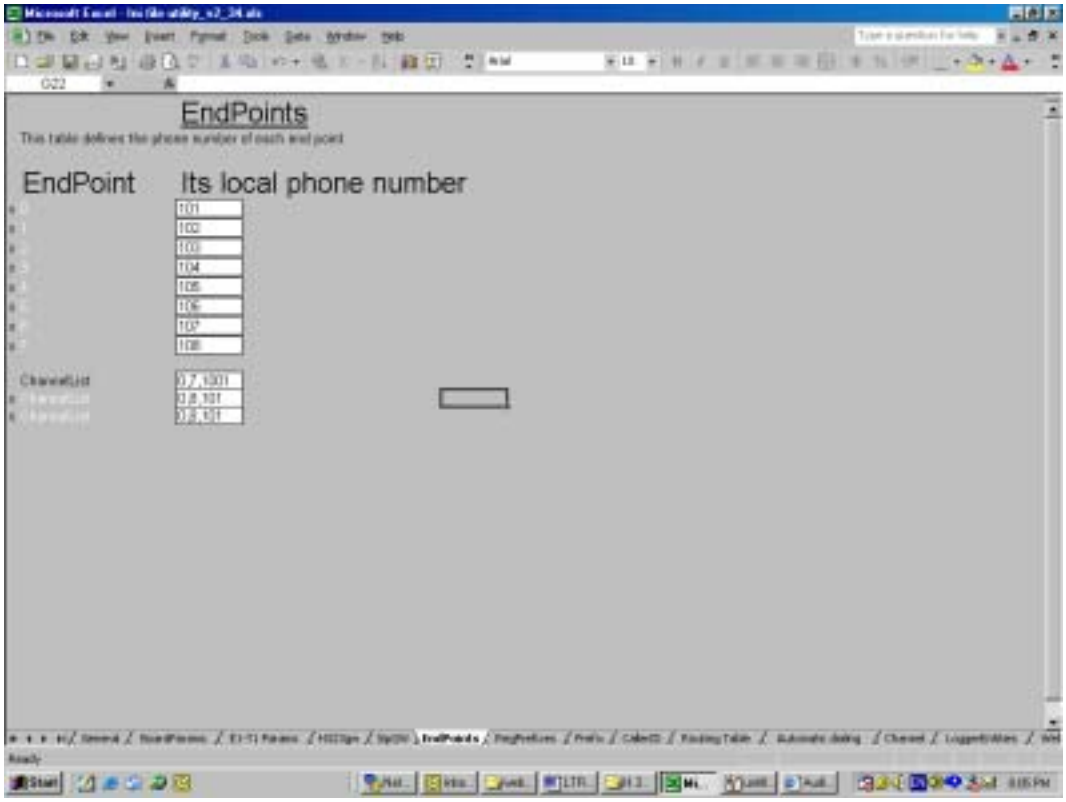
6.3.1 General Data Sheet

Figure 6-3: General Data Sheet



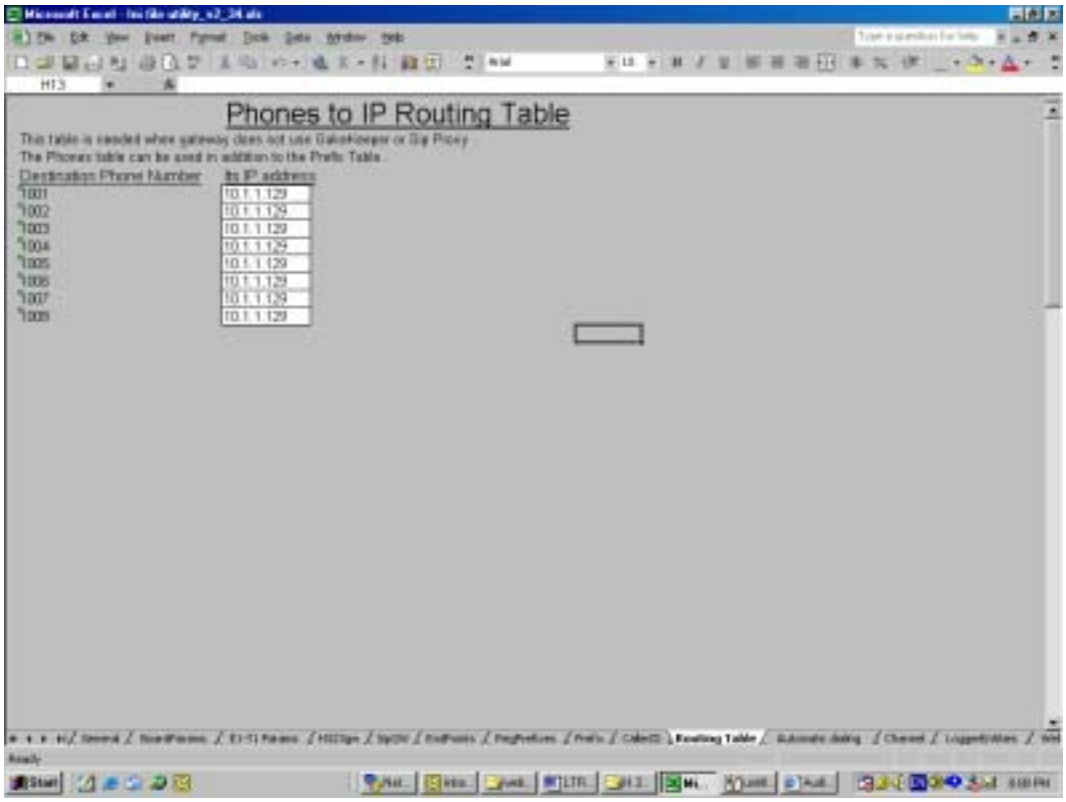
6.3.2 End Points Page

Figure 6-4: End Points Page



6.3.3 Phones to IP Routing Table

Figure 6-5: Phones to IP Routing Table



6.4 Using Call Progress Tones and Ringing

The Call Progress Tones Configuration File contains the definitions of the call progress tones and characteristics of Ringing signal to be detected/generated by the **MP-1xx**. Users can use either **MP-1xx,** one of the configuration files supplied by AudioCodes, or construct their own file.

The Call Progress Tones Configuration File used by the **MP-1xx** is a binary file (with the extension *dat*). Users can construct their own configuration file by starting from *tone.ini* file format, then modifying the file, and finally converting it into binary format using the "Download conversion utility" supplied with the **MP-1xx** package.

Please select "Convert dBm values" checkbox in the Conversion Utility.

To download the Call Progress Tones File to the **MP-1xx**, a correct definition should be used in the *mp108.ini* file. Refer to Section 6.4.4 for the description of the procedure on how to generate and download the Call Progress Tones file.

6.4.1 Format of the Call Progress *ini* File

The Call Progress Tones section of the *ini* file format starts from the following string:

[NUMBER OF CALL PROGRESS TONES] – containing only the following key: "Number of Call Progress Tones" defining the number of call progress tones to be defined in the file.

[CALL PROGRESS TONE #X] – containing the Xth tone definition (starting from 1 and not exceeding the number of call progress tones defined in the first section) using the following keys:

- **Tone Type –** Call Progress tone type

  1 - Dial Tone

  2 - Ringback Tone

  3 - Busy Tone

  4 - Congestion Tone

  5 - Special Information Tone

  6 - Warning Tone

  7 - Reorder Tone

  8 - Confirmation Tone

  9 - Call Waiting Tone

  16 - Off Hook Warning Tone

  23 - Hold Tone

- **Low Freq [Hz] –** Frequency in hertz of the lower tone component in case of dual frequency tone, or the frequency of the tone in case of single tone.
- **High Freq [Hz] –** Frequency in hertz of the higher tone component in case of dual frequency tone, or zero (0) in case of single tone.
- **Low Freq Level [-dBm] –** Generation level 0 dBm to –31 dBm in [dBm].
- **High Freq Level –** Generation level. 0 to –31 dBm. The value should be set to '32' in the case of a single tone.
- **First Signal On Time [10 msec] – "**Signal On" period (in 10 msec units) for the first cadence on-off cycle.
- **First Signal Off Time [10 msec] – "**Signal Off" period (in 10 msec units) for the first cadence on-off cycle.
- **Second Signal On Time [10 msec] – "**Signal On" period (in 10 msec units) for the second cadence on-off cycle.
- **Second Signal Off Time [10 msec] – "**Signal Off" period (in 10 msec units) for the second cadence on-off cycle.

Using this configuration file, the User can create up to 16 different call progress tones using up to 15 different frequencies (in the range of 300 Hz to 2000 Hz). Each one of the call progress tones is specified by the following two parameters: the tone frequency (either single or dual frequencies are supported) and the tone cadence. This is specified by 2 sets of ON/OFF periods, but Users can discard the use of the first On/Off cycle by setting the relevant parameters to zero. When the tone is made up of a single frequency, the second frequency field should be set to zero.

For a continuous tone (such as dial tone), only the "First Signal On time" should be specified. In this case, the parameter specifies the detection period. For example if it equals 300, then the tone is detected after 3 seconds (300 x 10 msec).

Note 1: When defining several continuous tones, the "First Signal On Time" parameter should have the same value for all tones.

Note 2: The tones frequency should differ by at least 40 Hz from one tone to other defined tones**.**

6.4.2 Default Template for Call Progress Tones

The **MP-1xx** is initialized with the default Call Progress Tones configuration template shown in Table 6-4. If you need to change one of the tones, edit the default call progress.txt file.

For example: to change the dial tone to 440 Hz only, replace the #Dial tone section in Table 6-4 with the following text:

#Dial tone

[CALL PROGRESS TONE #1]

Tone Type=1

Low Freq [Hz]=440

High Freq [Hz]=0

Low Freq Level [-dBm]=10 (-10 dBm)

High Freq Level [-dBm]=32 (use 32 only if a single tone is required)

First Signal On Time [10msec]=300; the dial tone is detected after 3 sec

First Signal Off Time [10msec]=0

Second Signal On Time [10msec]=0

Second Signal Off Time [10msec]=0

Users can specify several tones of the same type using Tone Type definition. These additional tones are used only for tone detection. Generation of specific tone is according to the first definition of the specific tone. For example, the User can define an additional dial tone by appending the second dial tone definition lines to the tone *ini* file. The **MP-1xx** reports dial tone detection if either one of the two tones has been detected.

Table 6-4: Call Progress Tones Template (continues on pages 74 to 77)

[NUMBER OF CALL PROGRESS TONES]
Number of Call Progress Tones=9
#Dial tone
[CALL PROGRESS TONE #0]
Tone Type=1
Low Freq [Hz]=350
High Freq [Hz]=440
Low Freq Level [-dBm]=13
High Freq Level [-dBm]=13
First Signal On Time [10msec]=300

Table 6-4: Call Progress Tones Template (continues on pages 74 to 77)

First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=0
Second Signal Off Time [10msec]=0
#Dial tone
[CALL PROGRESS TONE #1]
Tone Type=1
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=10
High Freq Level [-dBm]=32
First Signal On Time [10msec]=300
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=0
Second Signal Off Time [10msec]=0
#Ringback
[CALL PROGRESS TONE #2]
Tone Type=2
Low Freq [Hz]=440
High Freq [Hz]=480
Low Freq Level [-dBm]=19
High Freq Level [-dBm]=19
First Signal On Time [10msec]=0
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=200
Second Signal Off Time [10msec]=400
#Ringback
[CALL PROGRESS TONE #3]
Tone Type=2
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=16
High Freq Level [-dBm]=32
First Signal On Time [10msec]=0
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=100
Second Signal Off Time [10msec]=300
#Busy
[CALL PROGRESS TONE #4]

Table 6-4: Call Progress Tones Template (continues on pages 74 to 77)

Tone Type=3
Low Freq [Hz]=480
High Freq [Hz]=620
Low Freq Level [-dBm]=24
High Freq Level [-dBm]=24
First Signal On Time [10msec]=50
First Signal Off Time [10msec]=50
Second Signal On Time [10msec]=50
Second Signal Off Time [10msec]=50
#Busy
[CALL PROGRESS TONE #5]
Tone Type=3
Low Freq [Hz]=440
High Freq [Hz]=0
Low Freq Level [-dBm]=20
High Freq Level [-dBm]=32
First Signal On Time [10msec]=50
First Signal Off Time [10msec]=50
Second Signal On Time [10msec]=50
Second Signal Off Time [10msec]=50
#Reorder tone
[CALL PROGRESS TONE #6]
Tone Type=7
Low Freq [Hz]=480
High Freq [Hz]=620
Low Freq Level [-dBm]=24
High Freq Level [-dBm]=24
First Signal On Time [10msec]=25
First Signal Off Time [10msec]=25
Second Signal On Time [10msec]=25
Second Signal Off Time [10msec]=25
#Off Hook Warning Tone
[CALL PROGRESS TONE #7]
Tone Type=16
Low Freq [Hz]=1400
High Freq [Hz]=2600
Low Freq Level [-dBm]=0
High Freq Level [-dBm]=0

Table 6-4: Call Progress Tones Template (continues on pages 74 to 77)

First Signal On Time [10msec]=0
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=10
Second Signal Off Time [10msec]=10
#Hold Tone
[CALL PROGRESS TONE #8]
Tone Type=23
Low Freq [Hz]=1400
High Freq [Hz]=0
Low Freq Level [-dBm]=15
High Freq Level [-dBm]=32
First Signal On Time [10msec]=0
First Signal Off Time [10msec]=0
Second Signal On Time [10msec]=40
Second Signal Off Time [10msec]=500

6.4.3 Format of the Ringing Definition

The ringing pattern configures the ringing tone frequency and up to 4 ringing cadences. It is applicable for the **MP-1xx/FXS** Gateways. Only single ringing pattern can be defined, if not a default ringing pattern applies. The ringing frequency can be configured in the range from 10 Hz up to 200 Hz with a 5 Hz resolution. Each ringing cadence period can be defined as single ringing burst. Refer to the examples below.

The distinctive ringing section of the *ini* file format contains the following strings:

[NUMBER OF DISTINCTIVE RINGING PATTERNS]

- Number of Ringing patterns = 1
- [Ringing Pattern #0]
- Ring Type =0

- **Freq [Hz] –** Frequency in hertz of the ringing tone.
- **First Ring On Time [10 msec] – "**Ring On" period (in 10 msec units) for the first cadence on-off cycle.
- **First Ring Off Time [10 msec] – "**Ring Off" period (in 10 msec units) for the first cadence on-off cycle.
- **Second Ring On Time [10 msec] – "**Ring On" period (in 10 msec units) for the second cadence on-off cycle.
- **Second Ring Off Time [10 msec] – "**Ring Off" period (in 10 msec units) for the second cadence on-off cycle.
- **Third Ring On Time [10 msec] – "**Ring On" period (in 10 msec units) for the third cadence on-off cycle.
- **Third Ring Off Time [10 msec] – "**Ring Off" period (in 10 msec units) for the third cadence on-off cycle.

- **Fourth Ring On Time [10 msec] – "**Ring Off" period (in 10 msec units) for the forth cadence on-off cycle.
- **Fourth Ring Off Time [10 msec] – "**Ring Off" period (in 10 msec units) for the forth cadence on-off cycle.
- **Burst –** configures the ringing signal to be a single ringing burst comprised of all specified above cadences. The "Burst" string is defined per each ringing cadence and it must appear between "First/Second/Third/Forth" string and the "Ring On/Off Time".

6.4.3.1 Examples of Various Ringing Signals

#Regular North American Ringing Pattern: 20 Hz, 2 sec On, 4 sec Off
[NUMBER OF DISTINCTIVE RINGING PATTERNS]
Number of Ringing Patterns=1
[Ringing Pattern #0]
Ring Type=0
Freq [Hz]=20
First Ring On Time [10msec]=200
First Ring Off Time [10msec]=400

#GR-506-CORE Ringing Pattern 3: 20 Hz ringing comprised of three cadences
[NUMBER OF DISTINCTIVE RINGING PATTERNS]
Number of Ringing Patterns=1
[Ringing Pattern #0]
Ring Type=0
Freq [Hz]=20
First Ring On Time [10msec]=40
First Ring Off Time [10msec]=20
Second Ring On Time [10msec]=40
Second Ring Off Time [10msec]=20
Third Ring On Time [10msec]=80
Third Ring Off Time [10msec]=400

#EN 300 001 Ring – Finland: informative ringing nr. 3: three ringing bursts followed by repeated ringing of 1 sec on and 3 sec off.
[NUMBER OF DISTINCTIVE RINGING PATTERNS]
Number of Ringing Patterns=1
[Ringing Pattern #0]
Ring Type=0
Freq [Hz]=25
First Burst Ring On Time [10msec]=30
First Burst Ring Off Time [10msec]=30
Second Burst Ring On Time [10msec]=30
Second Burst Ring Off Time [10msec]=30
Third Burst Ring On Time [10msec]=30
Third Burst Ring Off Time [10msec]=30
Fourth Ring On Time [10msec]=100
Fourth Ring Off Time [10msec]=400

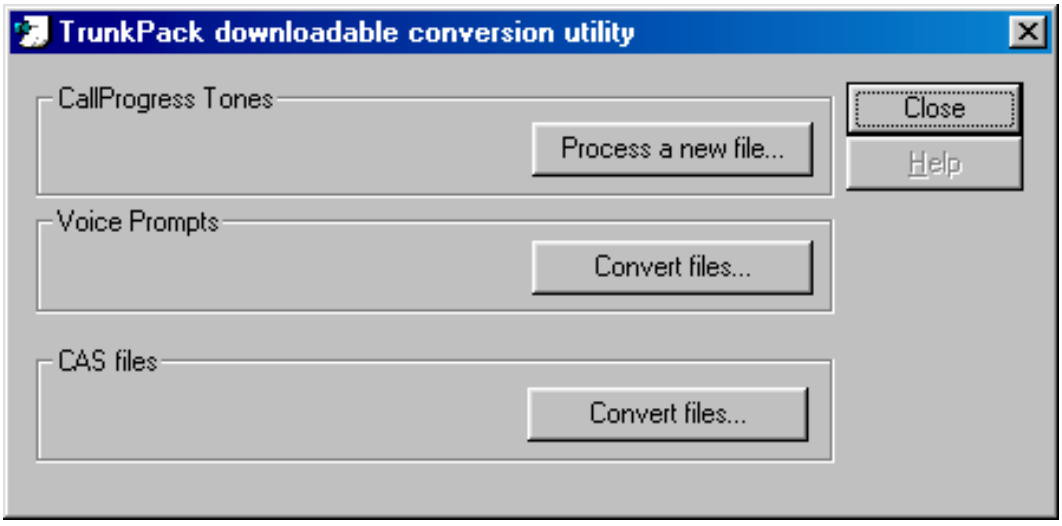
6.4.4 Call Progress Tone and Ringing Generation and Download Procedure

Follow the directions below for generation and download of the Call Progress Tone file.

➢ **To run the procedure take the following 10 steps:**

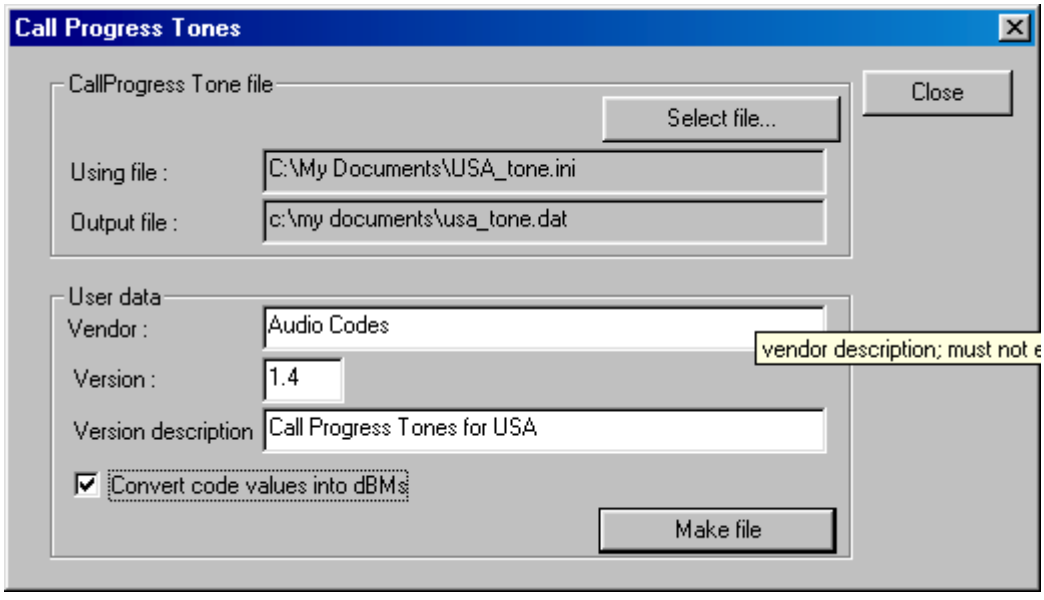
1. Prepare the ***tone.ini*** file including call progress tones and ringing parameters.
2. Use the "Download conversion utility" to generate binary ***tone.dat*** file

Figure 6-6: Download Selection Screen



3. Click "Process a new file.
4. Select input file such as ***usa_tone.ini*** and fill the Vendor and Version fields.

Figure 6-7: File Selection Screen



5. Select the 'Convert Code into dBm' checkbox
6. Click "Make File" button and then close the application.
7. Edit the ***mp-1xx.ini*** file and add the following two lines:

```
CallProgressTonesFilename = 'usa_tone.dat'
BurnCallProgressTonesFile = 1
```

8. Save the "***usa_tone.dat***" and "***mp108.ini***" files in TFTP folder
9. Set the Boot file name in the BootP server: ***mp108.ini***.
10. Activate the BootP and TFTP servers and reset the **MP-1xx** Gateway (refer to Section 6.1, describing **MP-1xx** provisioning).

6.5 The coeff.dat Configuration File

The purpose of the coeff.dat configuration file is to provide best feed and transmission quality adaptation for different phone line types. Two different *coeff.dat* files are needed for **MP-1xx/FXS** and for **MP-10x/FXO** Gateways. The file consists of a set of parameters for the signal processor of the loop interface devices. This parameter set provides control of the following AC and DC interface parameters:

- DC (battery) feed characteristics
- AC impedance matching
- Transmit gain
- Receive gain
- Hybrid balance
- Frequency response in transmit and receive direction
- Hook thresholds
- Ringing generation and detection parameters

This means, for example, that changing impedance matching or hybrid balance requires no hardware modifications, so that a single device is able to meet requirements for different markets. The digital nature of the filters and gain stages also ensures high reliability, no drifts (over temperature or time) and simple variations between different line types.

The *coeff.dat* configuration file is produced by AudioCodes for each market after comprehensive performance analysis and testing, and can be modified on request. The current file supports US line type of 600 ohm AC impedance and 40 V RMS ringing voltage for REN = 2.

In future software releases, it will be expanded to consist of different sets of line parameters, which can be selected in the *ini* file, for each port.

To support different types of countries and markets, it is necessary to support loading of a new *Coefficients.ini* file. This file consist of AC and DC line parameters for the peripheral devices. This file is loaded into the **MP-1xx** using the TFTP, in the same way as for the *tones.dat* file.

7 Device Management

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS Gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO Gateways.

7.1 SNMP Management

7.1.1 SNMP Overview

SNMP (Simple Network Management Protocol) is a standard network-based client/server-based control protocol to manage devices in the Network. The client program (called the Network Manager) makes connections to a server program, called the SNMP Agent. The SNMP Agent, embedded on a remote network device, serves information to the Network Manager regarding the device's status. The database used by the Agent to retrieve information, is referred to as the SNMP Management Information Base (MIB), and is a standard set of statistical and control values. Apart from the standard MIBs documented in IETF's RFCs, SNMP additionally allows the usage of private MIBs, containing non-standard information set.

Directives, issued by the network manager client to an SNMP Agent, consist of the identifiers of SNMP variables (referred to as MIB object identifiers or MIB variables) along with instructions to either get the value for the identifier, or set the identifier to a new value.

The definitions of MIB variables supported by a particular agent are incorporated in descriptor files, written in Abstract Syntax Notation (ASN.1) format, made available to network management client programs so that they can become aware of MIB variables and their usage.

The **MP-1xx** contains an embedded SNMP Agent supporting both general network MIBs (such as the IP MIB), VoP-specific MIBs (such as RTP, MGCP, etc.) and a proprietary MIB (known also as AudioCodes MIB) enabling a deeper probe into the inter-working of the Gateway. All the supported MIBs files are supplied as part of the release.

7.1.2 SNMP Message Standard

Four types of SNMP messages are defined:

- "**Get**" Request that returns the value of a named object.
- "**Get-Next**" Request that returns the next name (and value) of the "next" object supported by a network device given a valid SNMP name.
- "**Set**" Request that sets a named object to a specific value.
- "**Trap**" Message generated asynchronously by network devices. It notifies the network manager of a problem apart from polling of the device.

Each of the following message types fulfills a particular requirement of network managers:

- **Get Request**: Specific values can be fetched via the "get" request to determine the performance and state of the device. Typically, many different values and parameters can be

determined via SNMP without the overhead associated with logging into the device, or establishing a TCP connection with the device.

- **Get Next Request**: Enables the SNMP standard network managers to "walk" through all SNMP values of a device (via the "get-next" request) to determine all names and values that the device supports. This is accomplished by beginning with the first SNMP object to be fetched, fetching the next name with a "get-next", and repeating this operation until an error is encountered (indicating that all MIB object names have been "walked").
- **Set Request**: The SNMP standard provides a method of effecting an action associated with a device (via the "set" request) to accomplish activities such as disabling interfaces, disconnecting Users, clearing registers, etc. This provides a way of configuring and controlling network devices via SNMP.
- **Trap Message**: The SNMP standard furnishes a mechanism by which devices can "reach out" to a network manager on their own (via the "trap" message) to notify the manager of a problem with the device. This typically requires each device on the network to be configured to issue SNMP traps to one or more network devices that are awaiting these traps. The Trap messages are send to SNMP Manager. The IP address of SNMP Manager is defined in the *ini* file or via Web browser (in Network Settings)

7.1.3 SNMP MIB Objects

The SNMP MIB is arranged in a tree-structured fashion, similar in many ways to a disk directory structure of files. The top level SNMP branch begins with the ISO "internet" directory, which contains four main branches:

- The "**mgmt**" SNMP branch contains the standard SNMP objects usually supported (at least in part) by all network devices.
- The "**private**" SNMP branch contains those "extended" SNMP objects defined by network equipment vendors.
- The "**experimental**" and "**directory**" SNMP branches, also defined within the "internet" root directory, are usually devoid of any meaningful data or objects.

The "tree" structure described above is an integral part of the SNMP standard, however the most pertinent parts of the tree are the "leaf" objects of the tree that provide actual management data regarding the device. Generally, SNMP leaf objects can be partitioned into two similar but slightly different types that reflect the organization of the tree structure:

- **Discrete MIB Objects**: Contain one precise piece of management data. These objects are often distinguished from "Table" items (below) by adding a ".0" (dot-zero) extension to their names. The operator must merely know the name of the object and no other information.
- **Table MIB Objects**: Contain multiple pieces of management data. These objects are distinguished from "Discrete" items (above) by requiring a "." (dot) extension to their names that uniquely distinguishes the particular value being referenced. The "." (dot) extension is the "instance" number of an SNMP object. In the case of "Discrete" objects, this instance number is zero. In the case of "Table" objects, this instance number is the index into the SNMP table. SNMP tables are special types of SNMP objects, which allow parallel arrays of information to be supported. Tables are distinguished from scalar objects, in that tables can grow without bounds. For example, SNMP defines the "ifDescr" object (as a standard SNMP object) that indicates the text description of each interface supported by a particular device. Since network devices can be configured with more than one interface, this object could only be represented as an array.

By convention, SNMP objects are always grouped in an "Entry" directory, within an object with a "Table" suffix. (The "ifDescr" object described above resides in the "ifEntry" directory contained in the "ifTable" directory).

7.1.4 SNMP Extensibility Feature

One of the principal components of any respectable SNMP manager is a "MIB Compiler" which allows new MIB objects to be added to the management system. When a MIB is compiled into an SNMP manager, the manager is made "aware" of new objects that are supported by agents on

the network. The concept is similar to adding a new schema to a database.

Typically, when a MIB is compiled into the system, the manager creates new folders or directories that correspond to the objects. These folders or directories can typically be viewed with a "MIB Browser", which is a traditional SNMP management tool incorporated into virtually all network management systems.

The act of compiling the MIB allows the manager to know about the special objects supported by the agent and access these objects as part of the standard object set.

7.1.5 MP-1xx Gateway Supported MIBs

The **MP-1xx** Gateway contains an embedded SNMP Agent supporting the following MIBs:

- **The Standard MIB (MIB-II)** - The various SNMP values in the standard MIB are defined in RFC 1213. The standard MIB includes various objects to measure and monitor IP activity, TCP activity, UDP activity, IP routes, TCP connections, interfaces, and general system description.
- **RTP MIB** - The RTP MIB is supported per the RFC 2959. It contains objects relevant to the RTP streams generated and terminated by the Gateway and to the RTCP information related to these streams.
- **AcBoard MIB** - This AudioCodes proprietary MIB contains objects related both to the configuration of the Gateway and channels as well as to run-time information. Through this MIB, the User can set up the Gateway configuration parameters, reset the Gateway, and monitor the Gateway's operational robustness and quality of service during run-time.

7.2 Web Management

7.2.1 Overview

The **MP-1xx Gateway** contains an Embedded Web Server to be used both for Gateway configuration, including downloading of configuration files, and for run-time monitoring. The Web Server can be accessed from any standard Web browser, such as Microsoft™ Internet Explorer, Netscape™ Navigator, etc. Specifically, Users can employ this facility to set up the Gateway configuration parameters needed to configure the Gateway. Users also have the option to reset the Gateway to apply the new set of parameters.

Access to the Embedded Web Server is controlled by protection and security mechanisms described below.

Note: The **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have identical functionality (the **MP-102** supports FXS only), except for the number of channels, and are referred to collectively in this manual as the **MP-1xx**.

7.2.2 Password Control

The Embedded Web Server is protected by a unique username-password combination. The first time a browser request is made, the User is requested to provide its username-password so that the User can obtain access. Subsequent requests are negotiated by the browser on behalf of the User, so that the User doesn't have to re-enter the username-password for each request, but the request is still authenticated.

An additional level of protection is obtained by a restriction that no more than three IP addresses can access the Embedded Web Server concurrently. With this approach, a fourth User is told that the server is busy, even if the correct username-password was provided.

7.2.2.1 The Embedded Web Server Username-Password

The default username-password for all Gateways is shown below:

- Username = "Admin"
- Password = "Admin"

Change the Web password using the "Advanced Configuration Menu > Change Password" selection and then follow the pop-up window directives. The password can be a maximum of 7 case-sensitive characters. The new password is active only after restarting the Gateway using the reset button of the Embedded Web Server. Otherwise, the "old" password is still active.

The User can reset the Web username-password (to the default values) using an *ini* file parameter called "ResetWebPassword". The Web password is automatically the default password.

7.2.3 Web Configuration

The Embedded Web Server can be configured using *ini* file parameters.

7.2.3.1 Read-only Mode

The Embedded Web Server can be initialized to "read-only mode" by setting the "DisableWebConfig = 1" *ini* file parameter (the default state is read-write mode). In this mode, all the Web pages are presented in read-only mode. By selecting this mode, the User disables the capability to modify the configuration data. In addition, the User does NOT have access to the "Change Password" page or to the "Reset" page. When the Gateway is controlled through PCI, the Embedded Web Server is always in read-only mode.

7.2.3.2 Disable/Enable Embedded Web Server

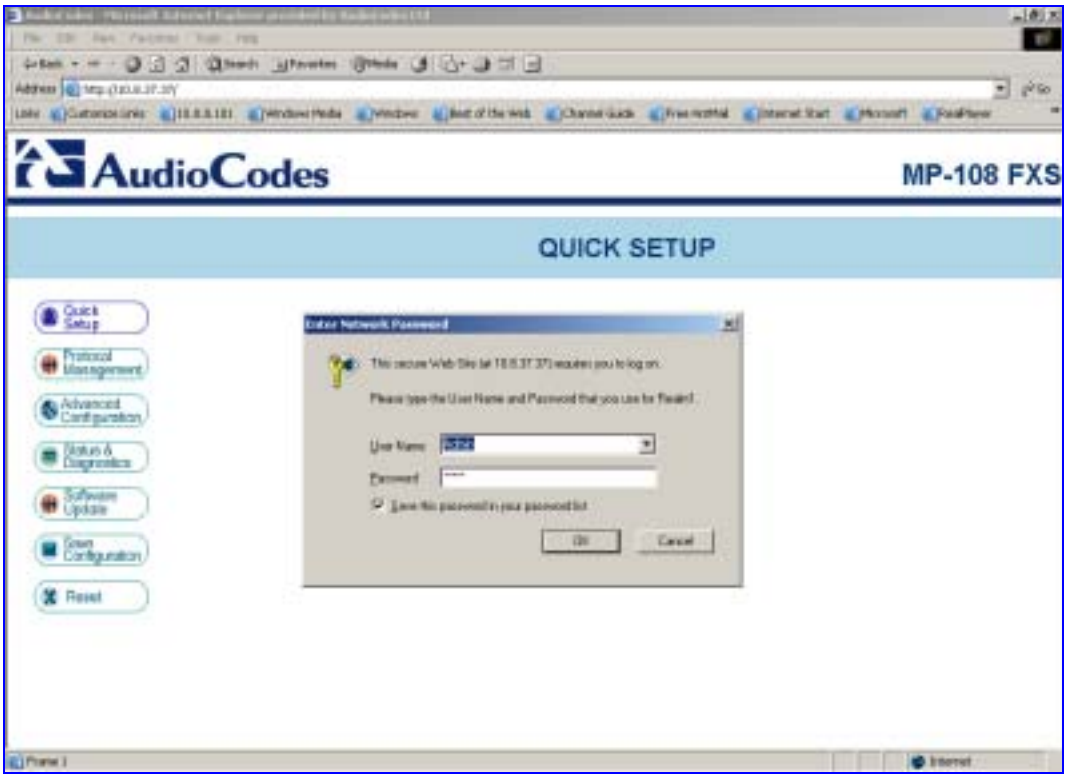
To deny access to the Gateway through HTTP protocol, the User has the capability to disable the Embedded Web Server task. To disable the Web task, use an *ini* file parameter called "DisableWebTask = 1". The default is to Web task enabled. When the Gateway is controlled

through PCI, the Embedded Web Server is always activated.

7.2.4 Using the Embedded Web Server

This section explains how to use the Embedded Web Server. After the initial IP address is set to the Gateway, you can connect with the integral Web-based configuration application. To access this Web application, use any standard Web-browsing application such as Microsoft™ Internet Explorer (Ver. 5.0 and higher), Netscape™ Navigator (Ver. 7.0 and higher), etc., and specify the IP address of the Gateway in the address field; the Embedded Web Server 'Enter Network Password' screen appears, as shown in Figure 7-1. Enter the User Name and Password (default: Admin, Admin). Click OK; the 'Quick Setup' screen is accessed. To configure the Gateway parameters you can either use this menu or go directly to "Protocol Management" and "Advance Configuration" menus. 'Quick setup' provides a basic set of Gateway configuration settings. An example of the 'Quick Setup' configuration is described in Section 4.3 on page 30.

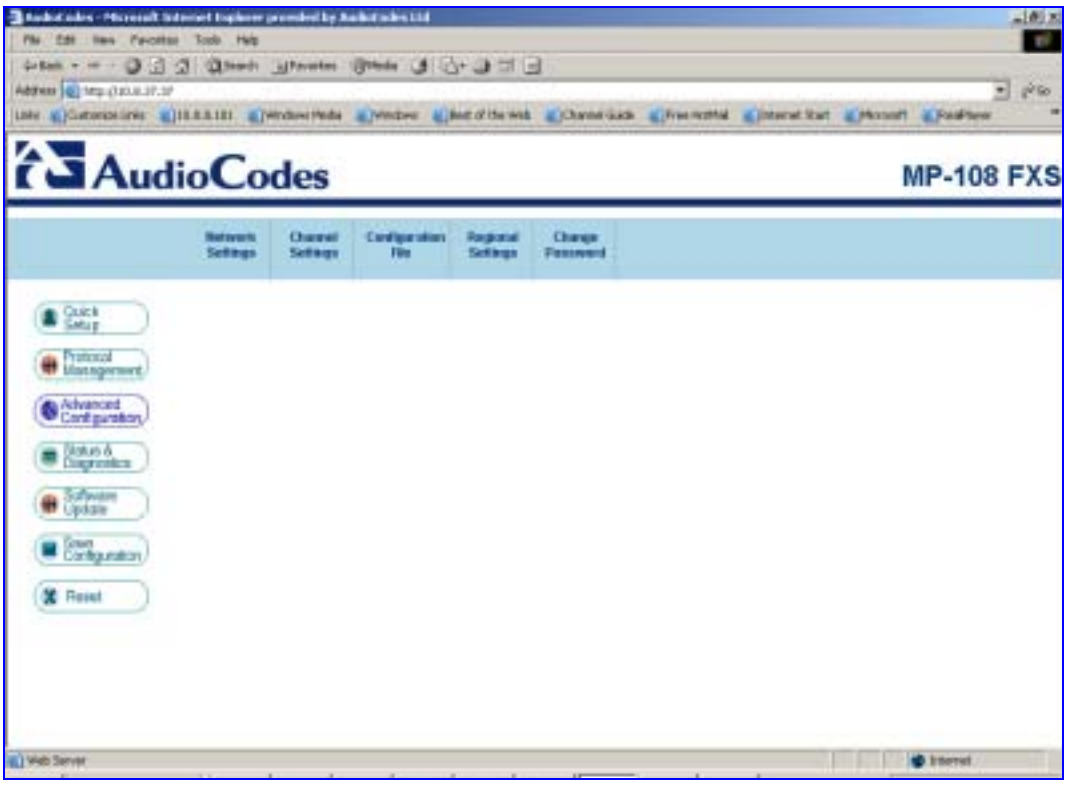
Figure 7-1: Embedded Web Server – Home Page



7.2.4.1 Set Up Gateway Configuration Parameters

Clicking the “Advanced Configuration” button leads to the following screen.

Figure 7-2: Embedded Web-Server - Gateway Parameters



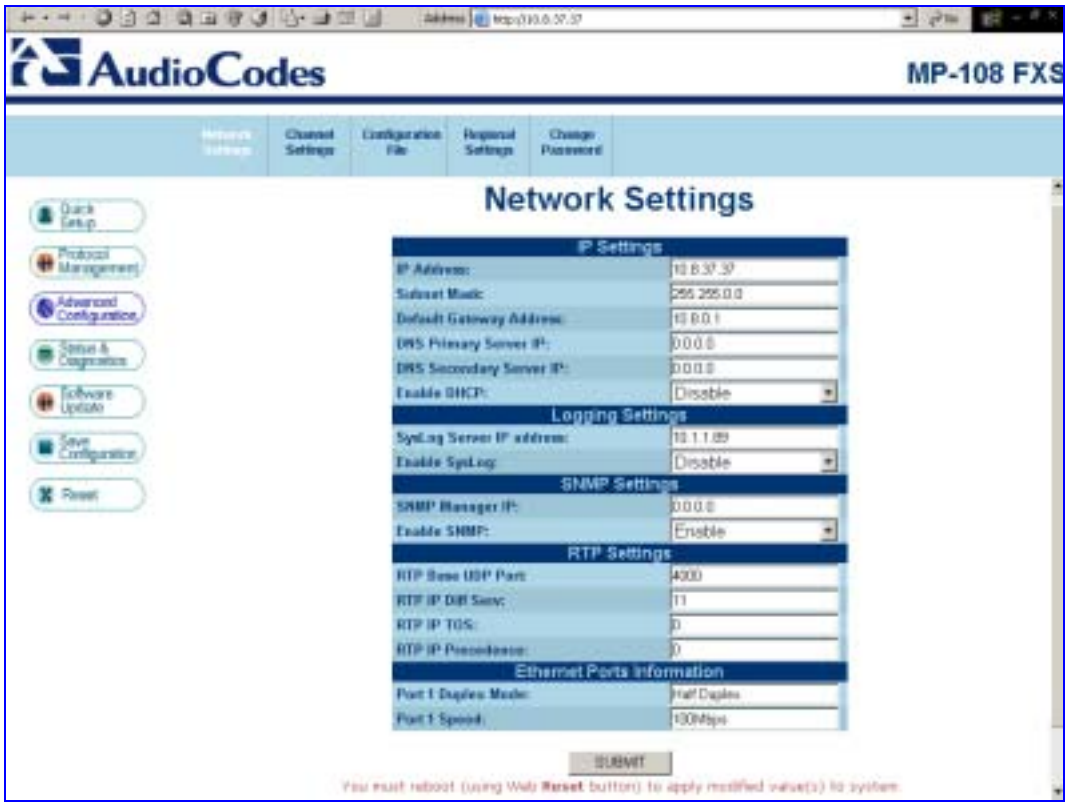
Selecting each of the sub menus shows the active configuration of each section and the values of the relevant parameters. From this menu you can select the following web pages:

- Network Settings
- Channel Settings
- Configuration File
- Regional Settings
- Change Password

7.2.4.2 Set up Gateway Network Parameters

To change the Gateway network settings, select the "Network Settings" tab shown below.

Figure 7-3: Web Server – Network Settings



From network settings page the User can define:

- IP settings including the Gateway IP address and Subnet Mask.
- Logging settings, such as IP address of SysLog Server. If the SysLog Server is disabled, the logging data is sent to the Gateway's serial RS-232 port.
- SNMP settings.
- RTP settings, including RTP Base UDP port, RTP IP Differentiated Services or RTP IP TOS and RTP IP Precedence QOS parameters.
- Ethernet status (read-only)

7.2.4.3 Channel Settings

From "Channel Settings" page the user can modify the following channel associated parameters:

- Voice Setting
- Fax/Modem/CID Settings
- RTP Settings
- IP Media Settings

7.2.4.4 Configuration File

Selecting the "Configuration Files" tab, enables the User to send new *ini* file to the Gateway or to upload from Gateway to PC the current *ini* file the Gateway is using. The uploaded *ini* file will include only parameters that were modified and contain other than default values.

Figure 7-4: Configuration Files

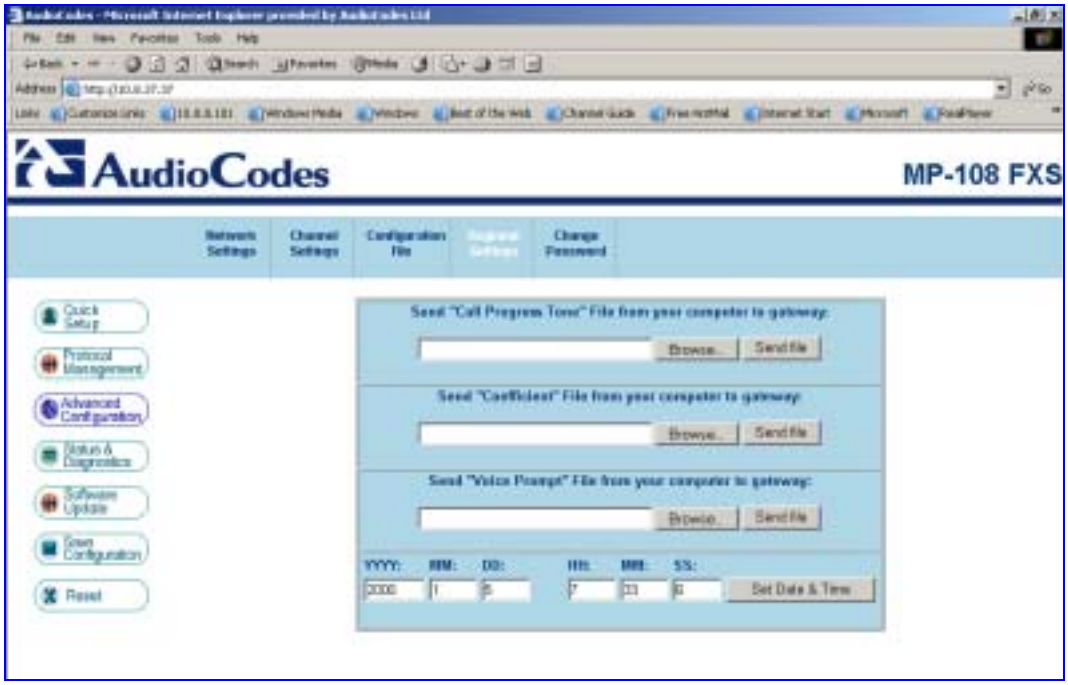
7.2.4.5 Regional Settings

From the "**Regional Settings"** page it is possible to download to the Gateway the following files:

- Call Progress Tone
- Coefficient file (different file for MP-1xx FXS and MP-10x FXO Gateways)
- Voice Prompt file – Currently not applicable for MP-1xx Gateways

In addition, you can set and view from this page the Gateway internal date and time.

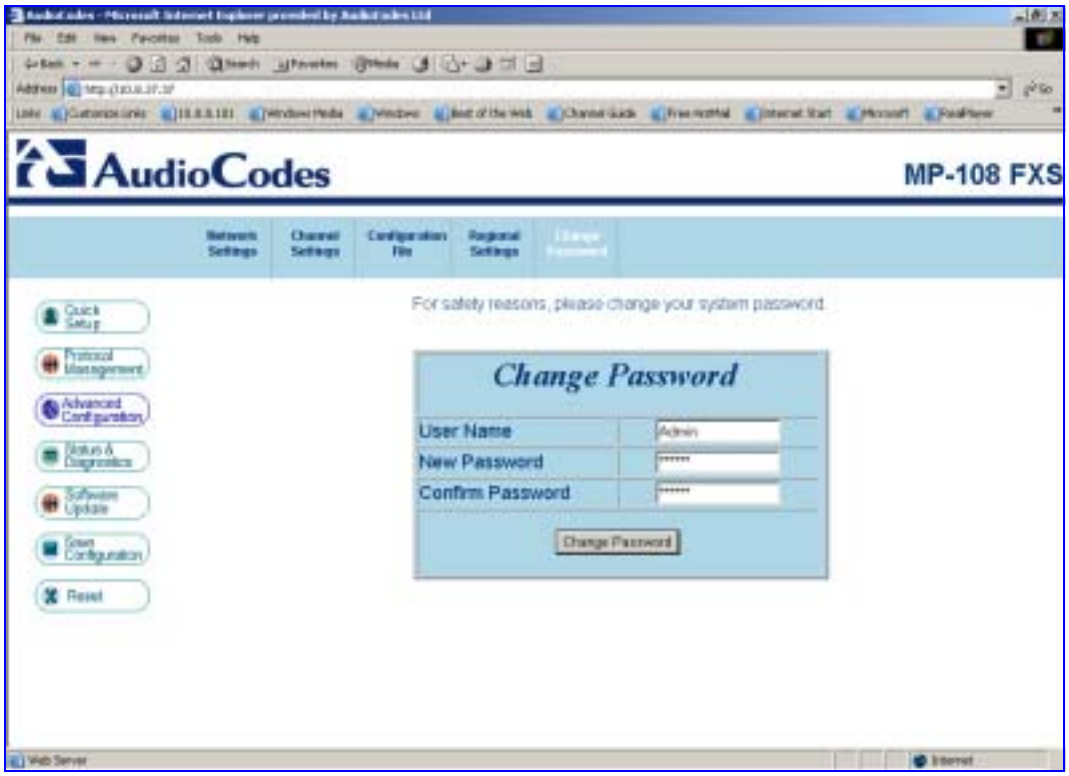
Figure 7-5: Regional Settings



7.2.4.6 Change Password

To change the User Name and Password select the "Change Password" tab as shown below.

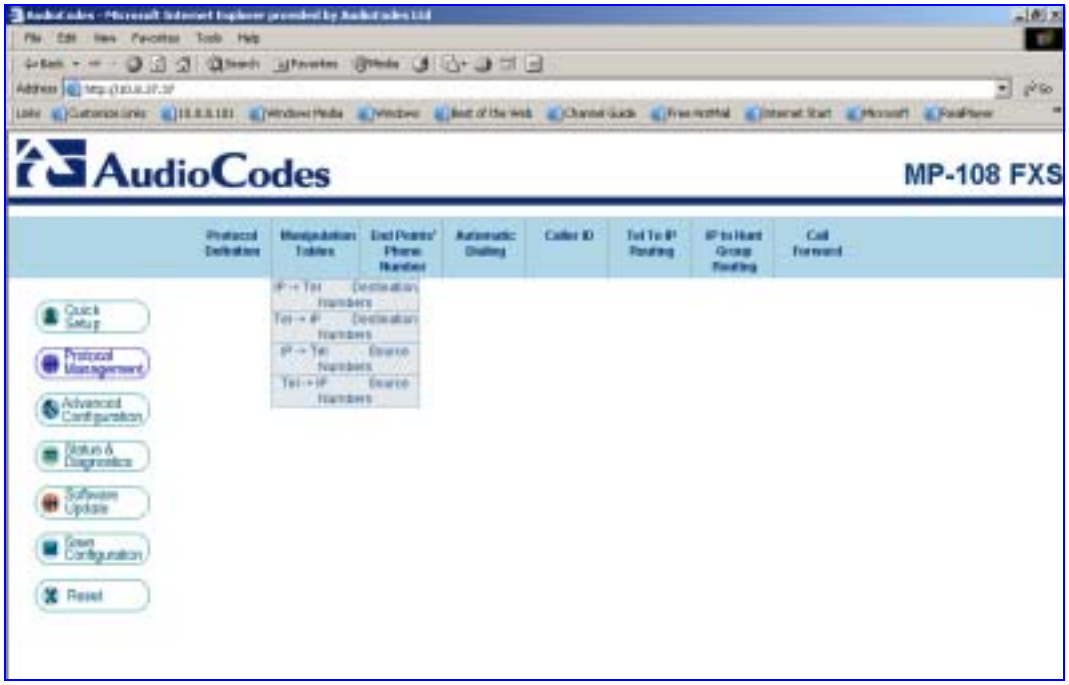
Figure 7-6: Web Server – Change Password



7.2.4.7 Set Up Gateway H.323 Parameters

To set the Gateway's H.323 parameters, select the "**Protocol Management"** tab shown below.

Figure 7-7: Web Server – Protocol Management Parameters



From this menu the User can select one of the following configuration pages:

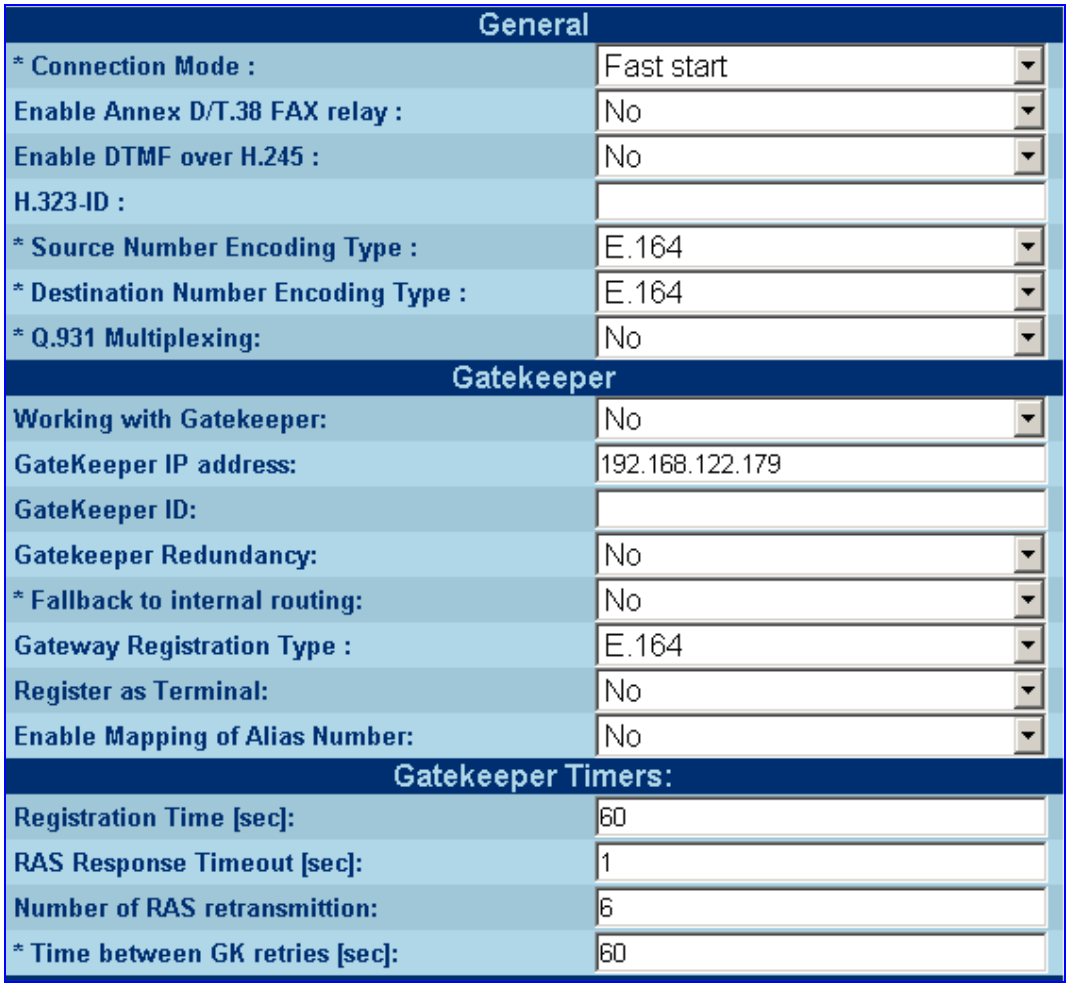
- Protocol Definition page, enabling to configure H.323 Gateway various parameters
- Number Manipulation tables
- Endpoints' Phone Numbers table
- Automatic Dialing table
- Caller ID table
- Tel to IP Routing table
- IP to Hunt Group Routing table
- Prefix Registration table (for MP-10x FXO Gateways)
- Call Forward table (For MP-1xx FXS Gateways)

7.2.4.8 Set Up H.323 Protocol Definition Parameters

"Protocol Definition" page enables the User to define H.323 configuration parameters. The H.323 protocol definition page includes several sections:

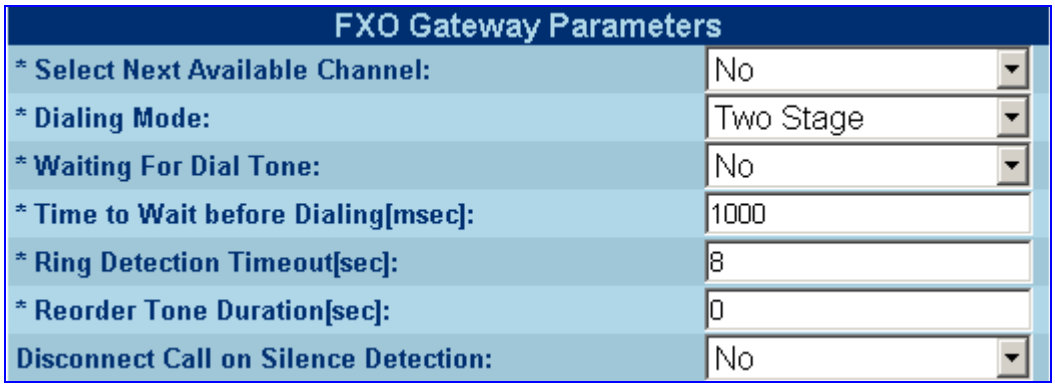
- General and Gatekeeper Parameters shown in Figure 7-8.

Figure 7-8: General and Gatekeeper Parameters


General
* Connection Mode : Fast start
Enable Annex D/T.38 FAX relay : No
Enable DTMF over H.245 : No
H.323-ID :
* Source Number Encoding Type : E.164
* Destination Number Encoding Type : E.164
* Q.931 Multiplexing: No
Gatekeeper
Working with Gatekeeper: No
GateKeeper IP address: 192.168.122.179
GateKeeper ID:
Gatekeeper Redundancy: No
* Fallback to internal routing: No
Gateway Registration Type : E.164
Register as Terminal: No
Enable Mapping of Alias Number: No
Gatekeeper Timers:
Registration Time [sec]: 60
RAS Response Timeout [sec]: 1
Number of RAS retransmittion: 6
* Time between GK retries [sec]: 60

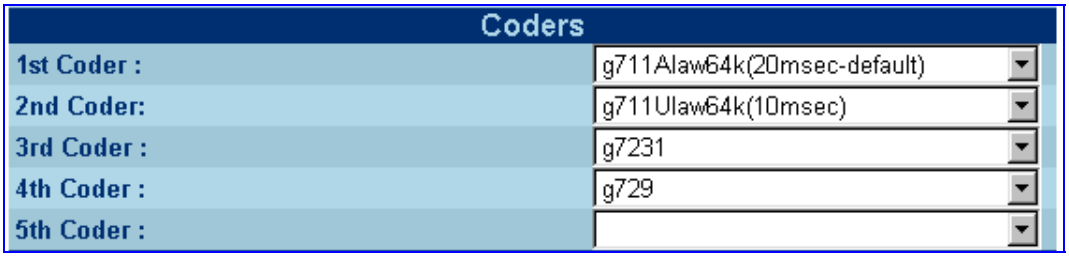
- FXO Gateway Parameters (Applicable only for MP-1xx/FXO Gateway) shown below in Figure 7-9.

Figure 7-9: FXO Gateway Parameters


FXO Gateway Parameters
* Select Next Available Channel: No
* Dialing Mode: Two Stage
* Waiting For Dial Tone: No
* Time to Wait before Dialing[msec]: 1000
* Ring Detection Timeout[sec]: 8
* Reorder Tone Duration[sec]: 0
Disconnect Call on Silence Detection: No

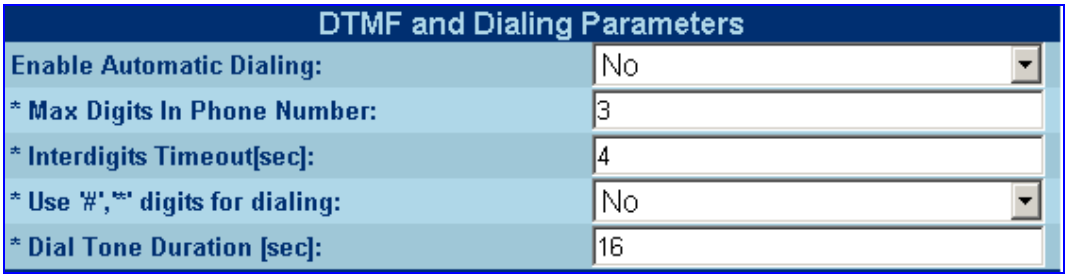
- VoIP Coders shown below in Figure 7-10.

Figure 7-10: Coders



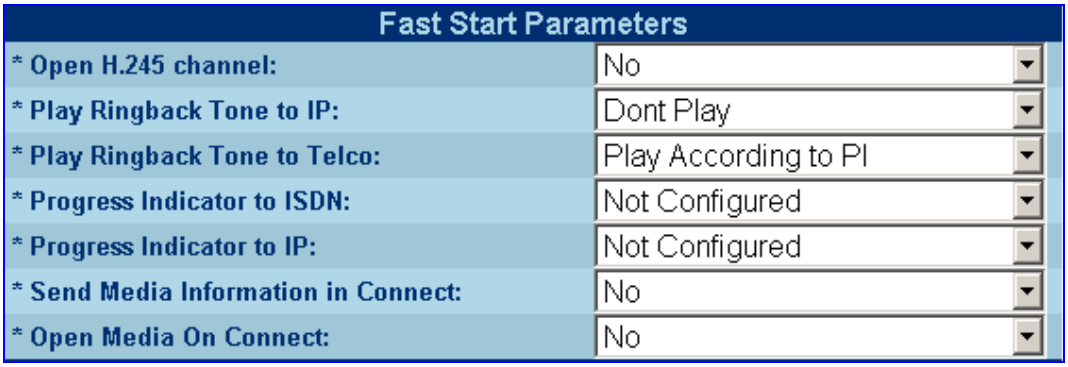
- DTMF and Dialing Parameters shown below in Figure 7-11.

Figure 7-11: DTMF and Dialing Parameters


DTMF and Dialing Parameters
Enable Automatic Dialing: No
* Max Digits In Phone Number: 3
* Interdigits Timeout[sec]: 4
* Use '#','*' digits for dialing: No
* Dial Tone Duration [sec]: 16

- Fast Start Parameters shown below in Figure 7-12

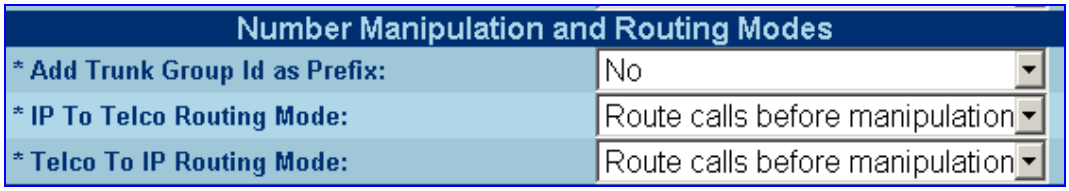
Figure 7-12: Fast Start Parameters


Fast Start Parameters
* Open H.245 channel: No
* Play Ringback Tone to IP: Dont Play
* Play Ringback Tone to Telco: Play According to PI
* Progress Indicator to ISDN: Not Configured
* Progress Indicator to IP: Not Configured
* Send Media Information in Connect: No
* Open Media On Connect: No

- Number Manipulation and Routing Modes Parameters. "Add Hunt Group ID as Prefix" feature adds the Hunt Group ID to destination number for Tel → IP calls. The routing and number manipulation rules can then be applied to the new number, enabling users to define routing rules based on the hunt group from where the call arrived. The two routing

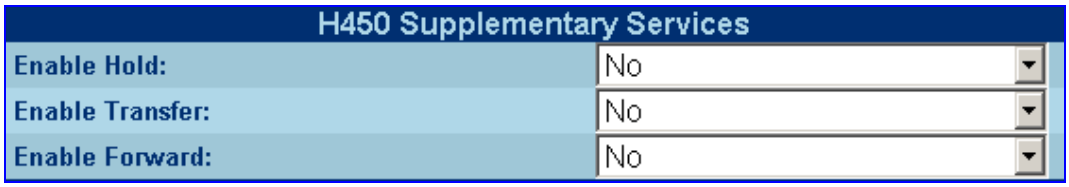
mode parameters define the execution order of number manipulation as opposed to the routing rules shown below in Figure 7-13.

Figure 7-13: Number of Manipulation and Routing Modes


Number Manipulation and Routing Modes
* Add Trunk Group Id as Prefix:
No
* IP To Telco Routing Mode:
Route calls before manipulation
* Telco To IP Routing Mode:
Route calls before manipulation

- H.450 Supplementary Services Parameters. These parameters enable Call Hold/Unhold, Call Transfer and Call Forwarding services shown below in Figure 7-14.

Figure 7-14: H.450 Supplementary Services



- Other Miscellaneous parameters shown below in Figure 7-15.

Figure 7-15: Miscellaneous Parameters

7.2.4.9 Configuration of Number Manipulation Tables

"Manipulation Tables" tab defines four manipulation tables used to modify destination and source numbers in IP→ Tel and Tel→ IP directions. Manipulation table for IP → Tel calls is shown below in Figure 7-16.

Figure 7-16: Phone Number Manipulation Table for IP → Tel calls

	Prefix	Num of stripped digits	Prefix to add	Number of digits to leave
1	03	0	972	
2	1001	4	5	
3	123451001	0	8	4
4				
5				
6				
7				
8				
9				
10				

In Figure 7-16 above, the 1st rule adds "972" prefix to all numbers starting with "03". The 2nd rule first deletes 4 digits from all numbers starting with "1001" and then the digit "5" is added as prefix. The 3rd rule deletes all digits except for the last four, from all numbers starting with 123451001 and adds the digit "8" as a prefix.

For example, number 035000 is changed to 972035000, number 1001876 is changed to 5876 and number 1234510012001 is changed to 82001.

7.2.4.10 Set Up Gateway Endpoints' Phone Numbers

"**Endpoint's Phone Numbers"** table is used to allocate phone numbers to Gateway ports, and to enable/disable Gateway ports. The table defines phone numbers for Gateway endpoints. The endpoints that aren't defined are disabled. In Channel(s) field a range of endpoints can be entered, such as "1-8" for **MP-108**, or "1-24" for **MP-124**. For a single endpoint, a single number can be entered in the channel(s) field. Hunt Group ID can optionally be configured to define a group of channels, which will be used for routing IP → Tel calls with common rules. Figure 7-17 on the next page, shows the Endpoint's Phone Numbers table.

Figure 7-17: Endpoint's Phone Number Table

Channel(s)	Phone Number	Hunt Group Id
1-4	101	
5	201	
6	202	
7	203	
8	204	

7.2.4.11 Set Up Automatic Dialing Table

"Automatic Dialing Table" defines destination numbers for Tel → IP calls. It can be used if "Automatic Dialing" feature is enabled (IsDialNeeded = 1). The table is applicable for FXS and FXO analog Gateways, for outgoing, Tel → IP calls. The table contains pre configured phone numbers per Gateway port. The number is automatically dialed if phone is picked up (for FXS

Gateway) or ring signal is applied to FXO Gateway port, refer to Figure 7-18.

Figure 7-18: Automatic Dialing Table

GW Port	Destination Phone Number
Port 1	1002
Port 2	1003
Port 3	1004
Port 4	1101
Port 5	1102
Port 6	1105
Port 7	1234
Port 8	2222

7.2.4.12 Set Up of Caller ID Display Table

"Caller ID Display Info" table, shown in Figure 7-19 contains Caller ID display information per **MP-1xx FXS/FXO** Gateway port. This information is sent in H.225 Setup message to remote party, for Tel ➔ IP calls. Remote party can use this display information for caller identification. The caller ID string can contain up to 18 characters.

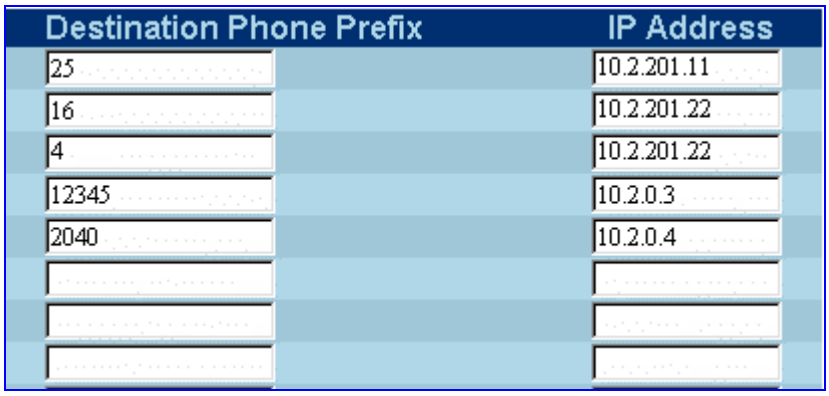
Figure 7-19: Caller ID Table

Gateway Port Number	Caller ID/Name
Port 1	Zohar
Port 2	Avi
Port 3	Michel
Port 4	Surpin
Port 5	Doron
Port 6	Tania
Port 7	Ilan
Port 8	Udi

7.2.4.13 Set Up of Tel to IP Routing Table

"**Tel to IP Routing & IP Security"** table is needed if the Gateway operates without a Gatekeeper. It contains up to 50 rows. Each row associates a called phone number prefix with destination IP address. The routing table can also be used for fallback routing if communication with Gatekeeper is lost. The IP addresses listed in the table can also be used for security. If security feature is enabled (SecureCallFromIP = 1) the Gateway will accept calls coming only from these addresses. The Phone to IP Routing Table is shown in Figure 7-20.

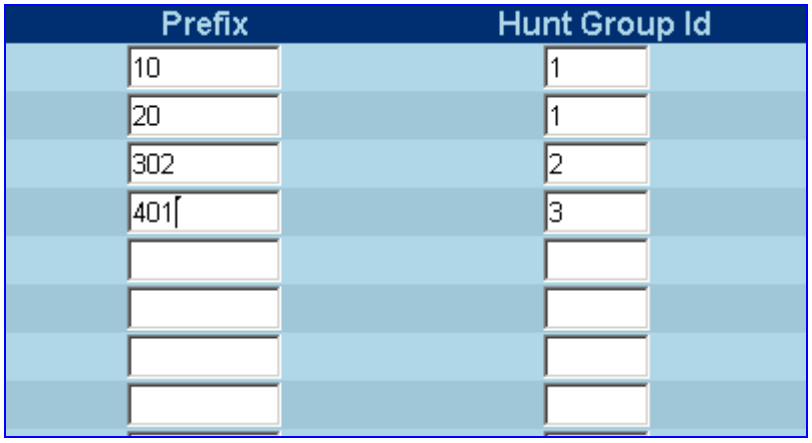
Figure 7-20: Phone to IP Routing & IP Security Table



7.2.4.14 Set Up of IP to Hunt Group Routing Table

"**IP to Hunt Group Routing" Table** can be used to route incoming IP calls to a group of channels. In the example below, IP incoming calls with called number starting with 10 or 20 will be routed to the 1st Hunt Group, while called numbers starting with 302 will be sent to the 2nd Hunt group. In each Hunt group next available Gateway port will be allocated. **In order to use Hunt Groups it is necessary to enable "Select Next Available Channel" in Protocol Definition web page (or configure IsUseFreeChannel = 1 in the *ini* file).** The IP to Hunt Group Routing Table is shown in Figure 7-21.

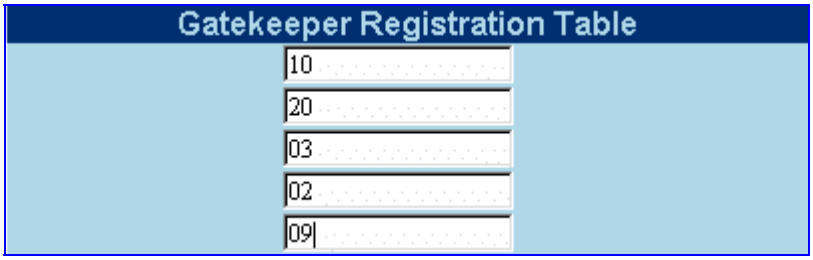
Figure 7-21: IP to Hunt Group Routing Table



7.2.4.15 Set Up of Registration Prefixes Table

"**Registration Prefixes Table**" – The registration prefixes are applicable for FXO Gateways for registration to Gatekeeper. Usually the prefixes define a range of PBX phone numbers that the Gateway can handle. If not used, the Gateway registers with its endpoint phone numbers. The Registration Prefixes Table is shown in Figure 7-22 below.

Figure 7-22: Registration Prefixes Table



7.2.4.16 Set Up of Call Forwarding Table

"**Call Forward"** table can be used for MP-1xx FXS Gateways, to define Call Forward Type per Gateway port. For example VoIP call, arriving to the Gateway's first port will be forwarded to destination number 201, if the endpoint is busy. Incoming IP→ Tel calls to Gateway ports 3 and 4 will be immediately forwarded to destination numbers 203 and 204 accordingly. The Call Forwarding table is shown in Figure 7-23.

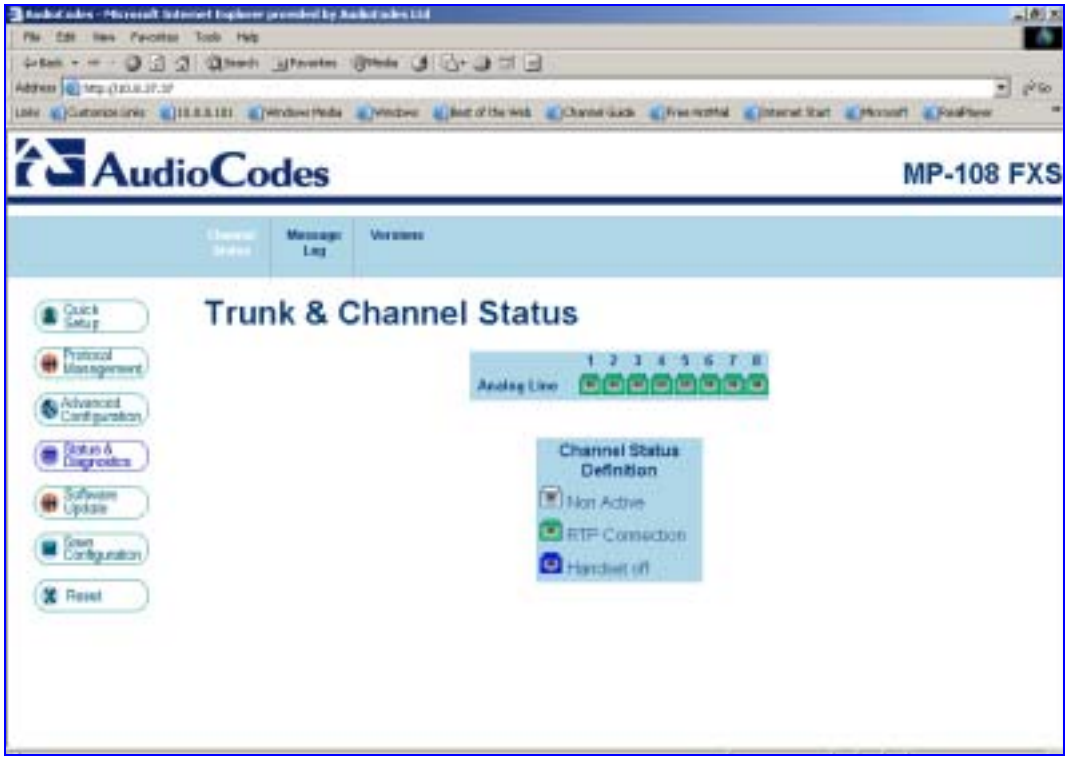
Figure 7-23: Call Forwarding Table

Gateway Port Number	Forward Type	Forward to Phone Number	Time for No Reply Forward
Port 1	on busy	201	0
Port 2	on busy	202	0
Port 3	immediate	203	0
Port 4	immediate	204	0
Port 5	no reply	205	20
Port 6	no reply	206	30
Port 7	not in use		0
Port 8	not in use		0

7.2.4.17 Channel Status Menu

Selecting the "**Status& Diagnostics"** tab and then the "Channel Status" page provides real time monitoring of the current channel status, as shown in the example below in Figure 7-24.

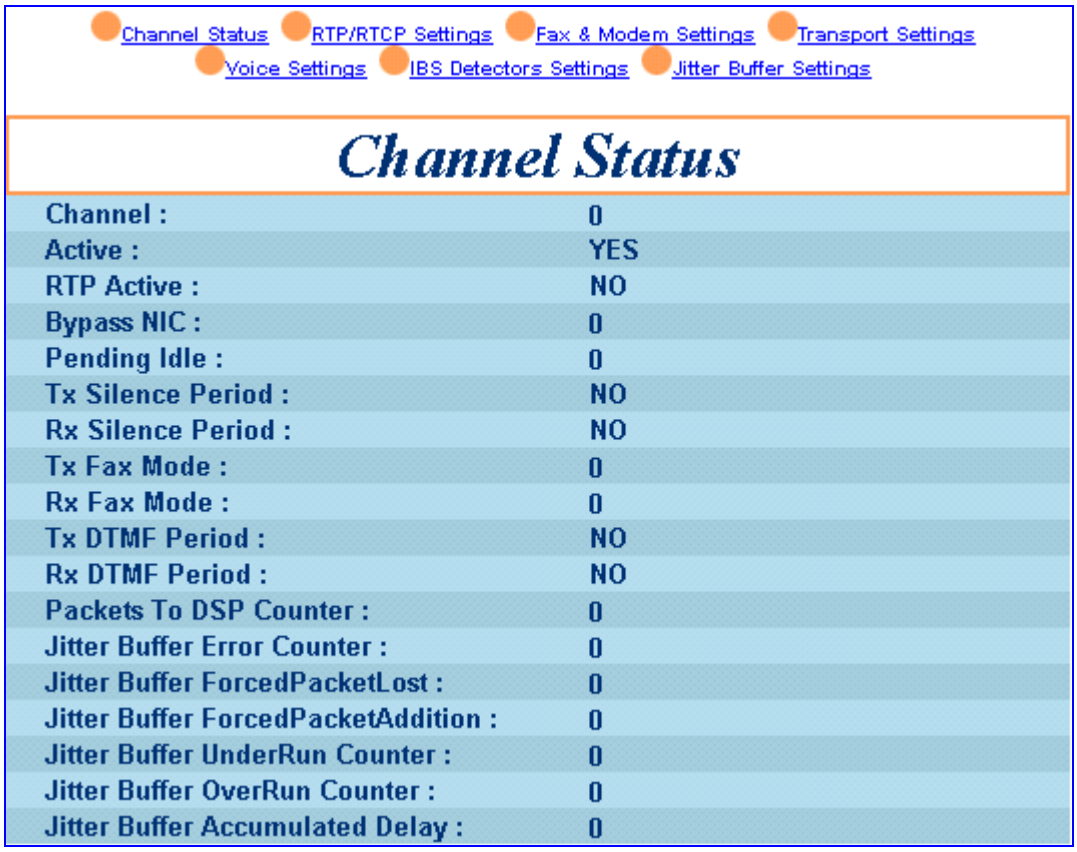
Figure 7-24: Channel Status



Active Channels are colored green.

To view the status of a specific channel, click on it; refer to Figure 7-25 on the following page.

Figure 7-25: Channel Status Detail


Channel Status
RTP/RTCP Settings
Fax & Modem Settings
Transport Settings
Voice Settings
IBS Detectors Settings
Jitter Buffer Settings
Channel Status
Channel : 0
Active : YES
RTP Active : NO
Bypass NIC : 0
Pending Idle : 0
Tx Silence Period : NO
Rx Silence Period : NO
Tx Fax Mode : 0
Rx Fax Mode : 0
Tx DTMF Period : NO
Rx DTMF Period : NO
Packets To DSP Counter : 0
Jitter Buffer Error Counter : 0
Jitter Buffer ForcedPacketLost : 0
Jitter Buffer ForcedPacketAddition : 0
Jitter Buffer UnderRun Counter : 0
Jitter Buffer OverRun Counter : 0
Jitter Buffer Accumulated Delay : 0

Selecting each of the sub menus shows the active channel configuration of each section and the values of its relevant parameters, as shown in the example above.

Reader's Notes

8 Diagnostics

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS Gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO Gateways.

8.1 Diagnostics Overview

AudioCodes provides several diagnostic tools to enable the User to identify an error condition and to provide a solution or work around when working with **MP-1xx** Gateway.

- LED Indication of channel activity status, data, control and LAN status.
- **MP-1xx** Self-Testing on hardware initialization.
- RS-232 terminal Notification Messages
- SysLog Event Notification Messages.
- Solutions to Common Problems.

They are described in the following pages.

8.2 MP-1xx Gateway Alarms & SNMP Traps

8.2.1 LED Visual Indicator Status and Alarms

Table 8-1: Indicator LEDs on the MP-1xx Front Panel

Label	Type	Color	State	Meaning
LAN	Ethernet Link Status	Green	ON	Valid Connection to 10/100 Base-T hub/switch
		Red	ON	Malfunction
Data	Packet Status	Green	Blinking	Transmitting RTP Packets
		Red	Blinking	Receiving RTP Packets
		Blank	-	No traffic
Control	Control Link	Green	Blinking	Sending and receiving H.323 messages.
		Red		Not supported in current release
Ready	Device Status	Green	ON	Device Powered, Self test OK
		Orange	Blinking	Software Loading/Initialization
		Red	ON	Malfunction

Table 8-2: MP-1xx Channel LEDs

MP-1xx with 1 to 8 Channels				
Label	**Type**	**Color**	**State**	**Meaning**
Activity	**FXS** Tel Port	Green	ON	Off-Hook/Ringing for Phone Port
	FXO Line Port	Green	ON	Line-Seize/Ringing State for Line Port

8.3 MP-1xx Self-Testing

The **MP-1xx** features two self-testing modes: rapid and detailed.

Rapid self-test mode is invoked each time the Media Gateway completes the initialization process. This is a short test phase in which the only error detected and reported is failure in initializing hardware components. All Status and Error reports in this self-test phase are reported through Network Interface ports, as well as indicated by the LED Status Indicators.

Detailed self-test mode is invoked when initialization of the Media Gateway is completed and if the configuration parameter EnableDiagnostics is set to 1 (this parameter can be configured through the *ini* file mechanism). In this mode, the Media Gateway tests all the hardware components (memory, DSP, etc.), outputs the status of the test results, and ends the test. To continue operational running, reset the Media Gateway again but this time configure the EnableDiagnostics parameter to 0.

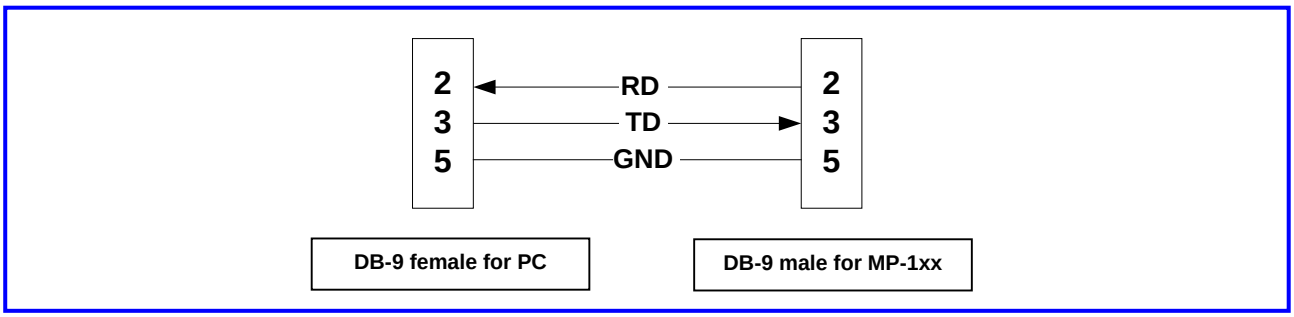
8.4 RS-232 Terminal

The **MP-1xx** status and error messages can be viewed via a terminal connected to the RS-232 management port.

➢ To connect MP-1xx to a HyperTerminal, take this step:

With a standard RS-232 straight cable (not a cross-over cable) with DB-9 connectors, connect the **MP-1xx** RS-232 port (it is marked "RS232") to either COM1 or COM2 RS-232 communication port on the PC. The connector pinout and gender are shown below in Figure 8-1.

Figure 8-1: RS-232 Cable Wiring



➢ To configure the HyperTerminal, take these 5 steps:

1. On a PC running a Windows™ operating system, open Start>Programs>Accessories>Communications>HyperTerminal; the Connection Description dialog opens.
2. Enter a name for the new connection in the Name field and click OK; the Connect To dialog opens.
3. In the Connect To dialog, enter COM1 or COM2, depending on the physical connection you performed when connecting the **MP-1xx** to the PC with the RS-232 cable; the COM1/2 Properties dialog opens.

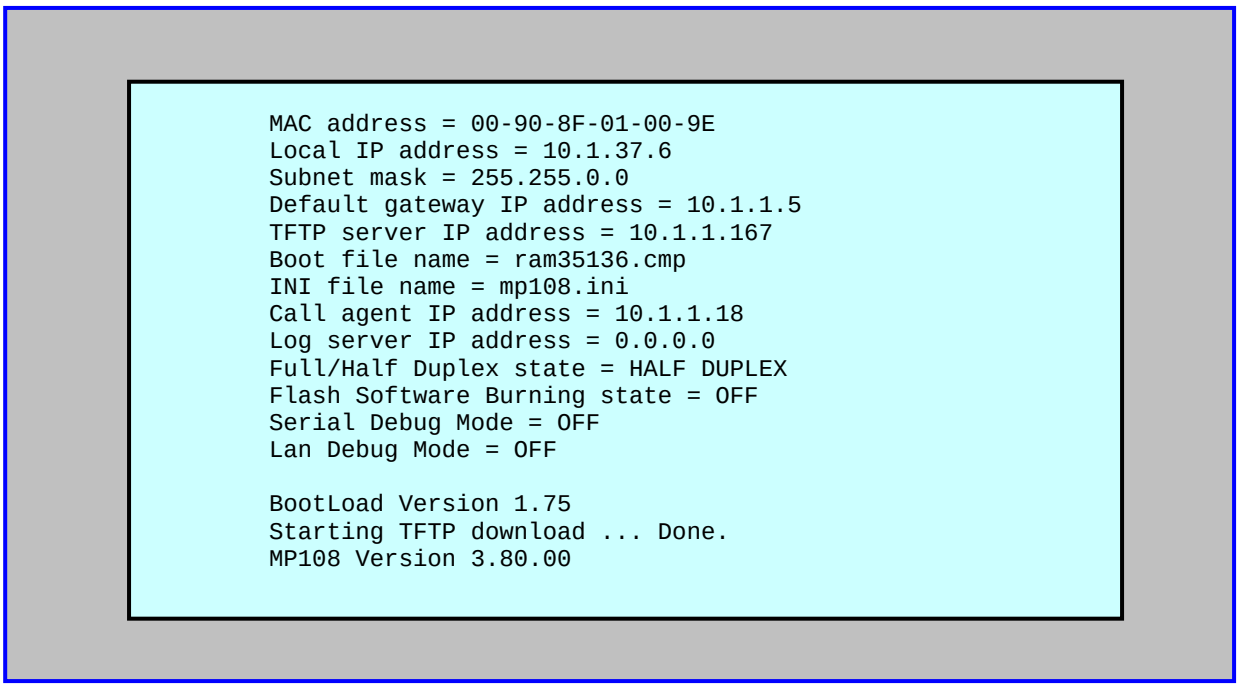
4. In the COM1/2 Properties dialog, enter the following settings for the serial communication port:

Baud Rate:	115,200 bps
Data bits:	8
Parity:	None
Stop bits:	1
Flow control:	Hardware

5. Click OK; the HyperTerminal main screen opens.

After applying power or reset the following information is printed on the terminal screen shown in Figure 8-2 on page 101. This information can be used to determine possible MP-1xx initialization problems. Such as incorrectly defined Local IP address, Subnet mask, Default router IP address, TFTP server IP address, BootFile name, *ini* file name and Full/Half duplex network state.

Figure 8-2: Status and Error Messages


MAC address = 00-90-8F-01-00-9E
Local IP address = 10.1.37.6
Subnet mask = 255.255.0.0
Default gateway IP address = 10.1.1.5
TFTP server IP address = 10.1.1.167
Boot file name = ram35136.cmp
INI file name = mp108.ini
Call agent IP address = 10.1.1.18
Log server IP address = 0.0.0.0
Full/Half Duplex state = HALF DUPLEX
Flash Software Burning state = OFF
Serial Debug Mode = OFF
Lan Debug Mode = OFF
BootLoad Version 1.75
Starting TFTP download ... Done.
MP108 Version 3.80.00

8.5 SysLog Support

8.5.1 Overview

SysLog protocol is an event notification protocol that allows a machine to send event notification messages across IP networks to event message collectors -also known as SysLog servers. The SysLog protocol is defined in RFC 3164 IETF standard.

Since each process, application and operating system was written somewhat independently, there is little uniformity to SysLog messages. For this reason, no assumption is made on the contents of the messages other than the minimum requirements of its priority.

SysLog uses UDP as its underlying transport layer mechanism. The UDP port that has been assigned to SysLog is 514.

The SysLog message is transmitted as an ASCII message. The message starts with a leading "<" ('less-than' character), followed by a number, which is followed by a ">" ('greater-than' character). This is optionally followed by a single ASCII space.

The number described above is known as the Priority and represents both the Facility and Severity as described below. The Priority number consists of one, two, or three decimal integers.

Example:

<37> Oct 11 16:00:15 mymachine su: 'su root' failed for lonvick on /dev/pts/8

8.5.2 SysLog Operation

8.5.2.1 Sending the SysLog Messages

AudioCodes' SysLog client, embedded in the firmware of the **MP-1xx**, sends error reports/events generated by the **MP-1xx** unit application to a SysLog server, using IP/UDP protocol. AudioCodes does NOT provide a SysLog server as several are provided as shareware that can be downloaded from the Internet.

Examples of SysLog Servers downloadable from the Internet:

1. Kiwi Enterprises: http://www.kiwi-enterprises.com/software_downloads.htm
2. The US CMS Server: http://uscms.fnal.gov/hanlon/uscms_server/
3. TriAction Software: http://www.triaction.nl/Products/SyslogDaemon.asp
4. Netal SL4NT 2.1 Syslog Daemon: *http://www.netal.com*

A typical SysLog server application enables filtering of the messages according to priority, IP sender address, time, date, …

8.5.2.2 Setting the SysLog Server IP Address

A SysLogServerIP Address parameter is supplied via Web browser or from an *ini* file in order to determine the address of the SysLog server.

8.5.2.3 Controlling the Activation of the SysLog Client

Activation of the SysLog client is controlled by an EnableSyslog *ini* file parameter. Setting it to 1 enables the SysLog protocol log.

8.5.2.4 The *ini* File Example for SysLog

Figure 8-3: The *ini* File Example for SysLog

```
[Syslog]
SyslogServerIP=10.2.0.136
EnableSyslog =1
LoggerFormat = 3
```

8.6 Solutions to Possible Problems

8.6.1 General

If there is a problem, check the following resources:

- Web browser status and channel parameter pages.
- Log messages of **MP-1xx** in HyperTerminal screen.
- BootP & TFTP log messages (for startup problems).
- Log messages in SysLog server.

8.6.2 Possible Common Problems

Possible common problems are described in Table 8-3.

Table 8-3: Possible Common Problems (continues on pages 103 to 103)

Problem	Possible Cause	What to do
No communication	Software does not function in **MP-1xx**	Try to "ping" to **MP-1xx**. If ping fails, check for network problems/definitions and try to reset the **MP-1xx**
	Network problem	Check cables.
	Network definitions	Check if default Gateway can reach IP of box.
		Check if box got the correct IP (it can be seen in the HyperTerminal screen).
		Check the validity of IP address, subnet and default Gateway. If default Gateway is not used, enter 0.0.0.0
	BootP didn't reply to box	Check if BootP server replied to **MP-1xx** at restart; it is seen in the BootP server log.
		Try to restart BootP server.
		Check the MAC address of the box in BootP server.
***ini* file was not loaded**	TFTP server down	Check if TFTP server working.
	TFTP server didn't get the request	Check this in its log.
	MP-1xx didn't request the file from your TFTP	Look in HyperTerminal for the TFTP server IP address that the **MP-1xx** is trying to use.
	TFTP server bug	Try to restart TFTP server.
	BootP sent to MP wrong TFTP server address	Check in HyperTerminal screen the address of used TFTP.
	ini file does not exists in default directory of TFTP	Check default directory of TFTP server and check that *ini* file exists there.
	Wrong *ini* file name	Verify in windows explorer that file extensions are displayed and the *ini* file isn't by mistake "XXX.ini.ini". Verify that extension *ini* is in lowercase letters.
	TFTP have too short timeout	Verify that: Timeout = 2 sec, # of retransmission = 10
Wrong *ini* file loaded	*ini* file is not in the correct position	Old *ini* file was probably loaded. Check which *ini* file was loaded. This can be done using HyperTerminal screen. The Gateway displays contents of *ini* file before it began.
	ini file corrupted	check *ini* file syntax
BootP reply from wrong BootP server	Other BootP servers contain MAC address of Gateway	Check that only your BootP server contains **MP-1xx** MAC address.

8.6.3 Possible H.323 Problems

Possible H.323 problems are described in Table 8-4.

Table 8-4: Possible H.323 Problems

Problem	Possible Cause	What to do
Gateway didn't register in Gatekeeper	The *ini* file was not updated to do so	Update *ini* file.
	No call was made	Gateway register only before first call.
Gatekeeper rejected registration	Gatekeeper already has same registered phone numbers	Change the Gateway phone numbers
G.711 voice quality is bad (clicks)	Silence compression not compatible (when working with a Gateway other than an AudioCodes Gateway)	Disable it and check if quality is better.
	Packet size not compatible (with G.711)	Check that packet period in remote side is 20 msec.
SETUP is sent before all digits were dialed	"MaxDigits" parameter has too small a number	Update the parameter.
No voice	There is no match in codecs	Check log in HyperTerminal. Change codec definition.

9 Specifications

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS Gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO Gateways.

9.1 MP-1xx Specifications

Table 9-1: MP-1xx Functional Specifications (continues on pages 105 to 107)

MP-1xx FXS Functionality	
FXS Capabilities	Short or Long Haul up to 3,000 m (10,000 ft) using 24 AWG line cord.
	Includes lightning and high voltage protection for outdoor operation.
	Caller ID generation: Bellcore GR-30-CORE Type 1 using Bell 202 FSK modulation or ETSI Type 1 (between first and second rings).
	Programmable Line Characteristics: Battery feed, line current, hook thresholds, AC impedance matching, hybrid balance, Tx & Rx frequency response, Tx & Rx Gains.
	Programmable ringing signal. Up to three cadences and frequency 15 to 200 Hz.
	Drive up to 4 phones per port (total 32 phones) simultaneously in Off-hook and Ring states. **MP-124** REN = 2 **MP-10x** REN = 5
	Over-temperature protection for abnormal situations as shorted lines.
	Loop-backs for testing and maintenance.
MP-10x FXO Functionality	
FXO Capabilities (does not apply to **MP-102**)	Short or Long Haul up to 7,000 m (24,000 ft) using 24 AWG line cord.
	Includes lightning and high voltage protection for outdoor operation.
	Programmable Line Characteristics: AC impedance matching, hybrid balance, Tx & Rx frequency response, Tx & Rx Gains, ring detection threshold, DC characteristics.
	Caller ID detection: Bellcore GR-30-CORE Type 1 using Bell 202 FSK modulation or ETSI Type 1 (between first and second rings).
Voice & Tone Characteristics	
Voice Compression	G.711 PCM at 64 kbps µ-law/A-law G.723.1 MP-MLQ at 5.3 or 6.3 kbps G.726 at 16 to 40 kbps ADPCM and E-ADPCM G.729A CS-ACELP at 8 kbps NetCoder at 6.4 to 8.8 kbps, 800-bit increments (proprietary coder)
Silence Suppression	G.723.1 Annex A G.729 Annex B PCM and ADPCM - Proprietary Voice Activity Detection (VAD) and Comfort Noise Generation (CNG) NetCoder

Table 9-1: MP-1xx Functional Specifications (continues on pages 105 to 107)

Echo Canceler	G.168, 25 msec
DTMF Transport	Mute, transfer in RTP payload or relay in compliance with RFC 2833
DTMF Detection and Generation	Dynamic range 0 to -25 dBm, compliant with TIA 464B and Bellcore TR-NWT-000506.
Call Progress Tone Detection and Generation	15 tones: single tone or dual tones, programmable frequency & amplitude; 16 frequencies in the range 300 to 2000 Hz, 1 or 2 cadences per tone, up to 2 sets of ON/OFF periods.
Output Gain Control	Programmable -31 dB to +31 dB in steps of 1 dB
Input Gain Control	Programmable -31 dB to +31 dB in steps of 1 dB
Fax/Modem Relay	
Fax Relay	Group 3 fax relay up to 14.4 kbps with auto fallback. T.38 compliant, real time fax relay Tolerant network delay (up to 9 sec round trip)
Modem Transparency	Auto switch to PCM or ADPCM on V.34 or V.90 modem detection
Protocols	
VoIP Signaling Protocol	H.323 ITU, Version 4
Communication Protocols	RTP/RTCP packetization. IP stack (UDP, TCP, RTP). Remote Software download (TFTP & BootP support).
Line Signaling Protocols	Loop start, FXS and FXO
Interfaces	
FXS Telephony Interface	2, 4, 8 or 24 Analog FXS phone or fax ports, loop start
FXO Telephony Interface	4 or 8 Analog FXO PSTN/PBX loop start ports
Network Interface	RJ-45 shielded connector, 10/100 Base-T.
RS-232 Interface	RS-232 Terminal Interface for maintenance and diagnostic reports. DB-9 connector on rear panel
Life Line (MP-10x/FXS)	Life Line, connected to the unused pins on port #4 (port #2 for **MP-102**/FXS), with a relay to an analog line, even if the **MP-10x/FXS** is powered off (refer to Section 2.1.5 for details). Does NOT function with **MP-124** and **MP-10x**/FXO Gateways.
Connectors & Switches	
Rear Panel	
24 Analog Lines (MP-124)	50-pin Telco shielded connector
8 Analog Lines (MP-108)	8 RJ-11 connectors
4 Analog Lines (MP-104)	4 RJ-11 connectors
2 Analog Lines (MP-102)	2 RJ-11 connectors
Ethernet	10/100 Base-T, RJ-45 shielded connector
RS-232	Console port - DB-9
Front Panel	
Reset	Resets the **MP-1xx**
Physical	
MP-10x Enclosure Dimensions	Width: 221 mm 8.7 in Height: 44.5 mm 1.75 in Depth: 240 mm 9.5 in Weight: 1.24 kg 2.5 lb
MP-124 Enclosure Dimensions	1U, 19-inch Rack Width: 445 mm 17.5 in Height: 44.5 mm 1.75 in Depth: 269 mm 10.6 in Weight: 2.24 kg 4.9 lb
Environmental	Operational: 0° to 45° C 32° to 113° F

Table 9-1: MP-1xx Functional Specifications (continues on pages 105 to 107)

	Storage: -10° to 70° C 14° to 158° F Humidity: 10 to 90% non-condensing
Installation	Desk-top, shelf, wall mount or 19-inch rack mount with side brackets.
Electrical	Universal 90-260 VAC, 1A, 47-63 Hz
Type Approvals	
Telecommunication	FCC part 68 & CE CTR21
Safety and EMC	UL 1950, FCC part 15 Class B CE Mark (EN 60950, EN 55022, EN 55024)
Management	
Configuration	Gateway configuration using Web browser, *ini* files or local RS-232 console
Management and Maintenance	SNMP
	SysLog, per RFC 3164
	Local RS-232 terminal
	Web Management

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10 Appendices

This section contains the following Appendices:

Appendix A – BootP/TFTP Configuration Utility	Error! Bookmark not defined.
Appendix B - RTP/RTCP Payload Types	**119**
Appendix C - DTMF, Fax & Modem Modes	**123**
Appendix D - DHCP Server Configuration	**127**

Note 1: The **MP-124** 24-port, **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Media Gateways have similar functionality except for the number of channels (the **MP-124** and **MP-102** support only FXS), and all versions are referred to collectively in these release notes as the **MP-1xx**.

Note 2: **MP-10x** refers to **MP-108** 8-port, **MP-104** 4-port and **MP-102** 2-port Gateways.

Note 3: **MP-1xx**/FXS refers only to the **MP-124**/FXS, **MP-108**/FXS, **MP-104**/FXS and **MP-102**/FXS Gateways.

Note 4: **MP-10x**/FXO refers only to **MP-108**/FXO and **MP-104**/FXO Gateways.

Appendix A - BootP/TFTP Configuration Utility

A.1 Introduction

The AudioCodes **BootP/TFTP Configuration Utility** enables easy configuration and provisioning of AudioCodes products. It contains BootP and TFTP servers with specific adaptations to AudioCodes' requirements.

A.1.1 Key Features

- Internal BootP server supporting hundreds of entities.
- Internal TFTP server.
- Contains all required data for AudioCodes products in predefined format.
- Provides a TFTP server address, enabling network separation of TFTP and BootP servers.
- Tools to backup and restore the local database.
- Templates.
- User-defined names for each entity.
- Option for changing MAC address.
- Protection against entering faulty information.
- Remote reset (for this version and above).
- Unicast BootP respond.
- User-initiated BootP respond, for remote provisioning over WAN.
- Filtered display of BootP requests.
- Location of other BootP servers that contain the same MAC entity.
- Common log window for both BootP and TFTP sessions.
- Works with Windows™ 98, Windows™ NT, Windows™ 2000.

A.1.2 Specifications

BootP standards: RFC 951 and RFC 1542
TFTP standards: RFC 1350 and RFC 906
Operating System: Windows™ 98, Windows™ NT and Windows™ 2000
Max number of MAC entries: 200

BootP Fields:

- Hardware address (MAC): 12 hex digits.
- IP address.
- Subnet.
- Default Gateway.
- TFTP server IP; (Using the TFTP server IP field enables the download of a software image from a different Host).
- Boot File.
- *ini* File.
- Call Agent IP.
- New MAC (optional).

Screens:

- Logging screen.
- Preferences screen.
- Client Configuration screen.
- Template definition screen.

A.1.3 BootP/TFTP Configuration Utility Installation

The BootP/TFTP Configuration Utility can be installed on a PC from the AudioCodes Software & Documentation CD:

TrunkPack Boards (TP)	Compact disk Catalog Number: LSTC-00001,
IPmedia Platforms (IPM)	Compact disk Catalog Number: LSTC-00002
TrunkPack Modules (TPM)	Compact disk Catalog Number: LSTC-00003
MediaPack Series (MP)	Compact disk Catalog Number: LSTC-00005

To install the BootP/TFTP Configuration Utility, unzip the TP3.810.exe file and navigate to the BootP *.exe file. The installation procedure is facilitated by prompts. After completing the procedure, open Start>Programs>BootP; the BootP/TFTP Server main screen is displayed.

A.1.4 Logging Screen

The AudioCodes BootP/TFTP Configuration Utility main screen (Figure 10-1 on page 114) includes the Log line, printed per BootP request with the following parameters:

- Hardware (MAC) address.
- Status (found or not found in cache).
- Date and Time.
- Assigned IP address (if found).
- Client name.

Clicking on Log line shows all BootP reply parameters or enables entry to a new entity.

A.1.5 Preferences Window

The Preferences Window (Figure 10-2 on page 115) is used to define BootP and TFTP configuration parameters:

- TFTP directory.
- *ini* File Mask.
- Boot File Mask.
- TFTP timeout and number of retransmissions.
- BootP replay type (broadcast or unicast).
- BootP ARP mode (dynamic or static).
- Number of initiated BootP replies (send after remote reset), optionally used when the Media Gateway (MP-102, MP-104, MP-108 and MP-124, is installed behind the firewall that blocks BootP broadcast requests.

A.1.6 Client Configuration Window

The Client Configuration Window (Figure 10-3 on page 116) shows:

- All client entities
- MAC
- Name
- IP per entity.

Using this screen, the User can:

- Add a new entry
- Delete an existing entry
- Modify an existing entry
- Test a selected client for finding all BootP servers that respond to a BootP request with a specific MAC address.

If a template is selected, any parameter can be entered manually or copied from the selected template, by marking the checkbox on the right side of the parameter. Usually, only an IP address is entered manually, while other parameters are copied from the template.

A.1.7 Template Window

The Template Window (Figure 10-4 on page 118) enables the User to add, modify, or delete templates.

The template includes:

- Subnet.
- Gateway, TFTP server.
- BootFile.
- *ini* file.
- Call Agent fields.
- Server IP.

A.2 Window Details

A.2.1 Main Window

Figure 10-1: Main Window

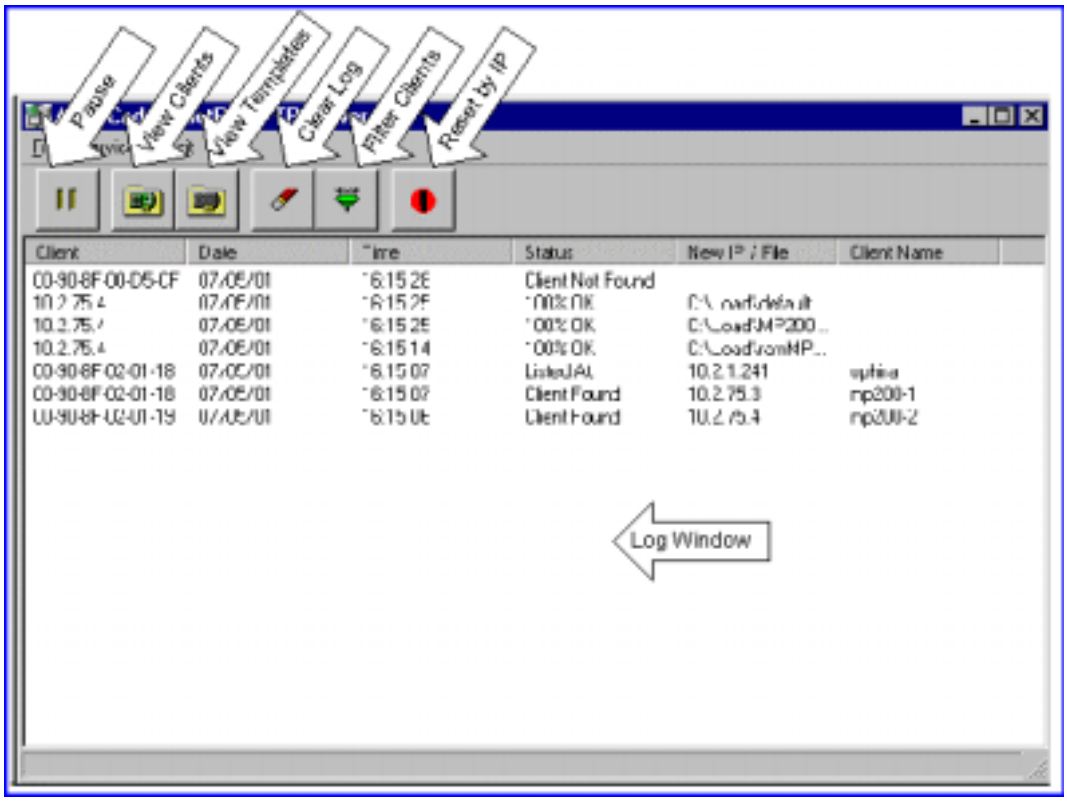


Figure 10-1 is the main window of the program. It features several controls, listed below:

Program State - With this button, the User can pause the program. When the program has paused, no replies to BootP requests are sent.

View Clients Button - This button opens up the Clients Configuration window.

View Templates Button - This button opens up the Templates Configuration window.

Clear Log Button - This buttons clears the log.

Filter Unknown Clients Button - This button filters all BootP requests that are not in the client configuration list.

Reset Button - This button opens up a dialog box, where the User can enter an IP of a client. The program sends a reset command to that client.

Edit/Preferences - Selecting “Preferences” from the Edit menu opens the Preferences window for defining BootP and TFTP parameters.

Log Window - All BootP requests and TFTP sessions are displayed, including the time and date of the request. In addition, the response type is also displayed:

- “Client Not Found”
- “Client Found”
- “Client’s Mac Changed”
- “Client Disabled”
- “Listed at” (when using the “test selected clients” button).
- For TFTP session, File name and Download status are displayed.

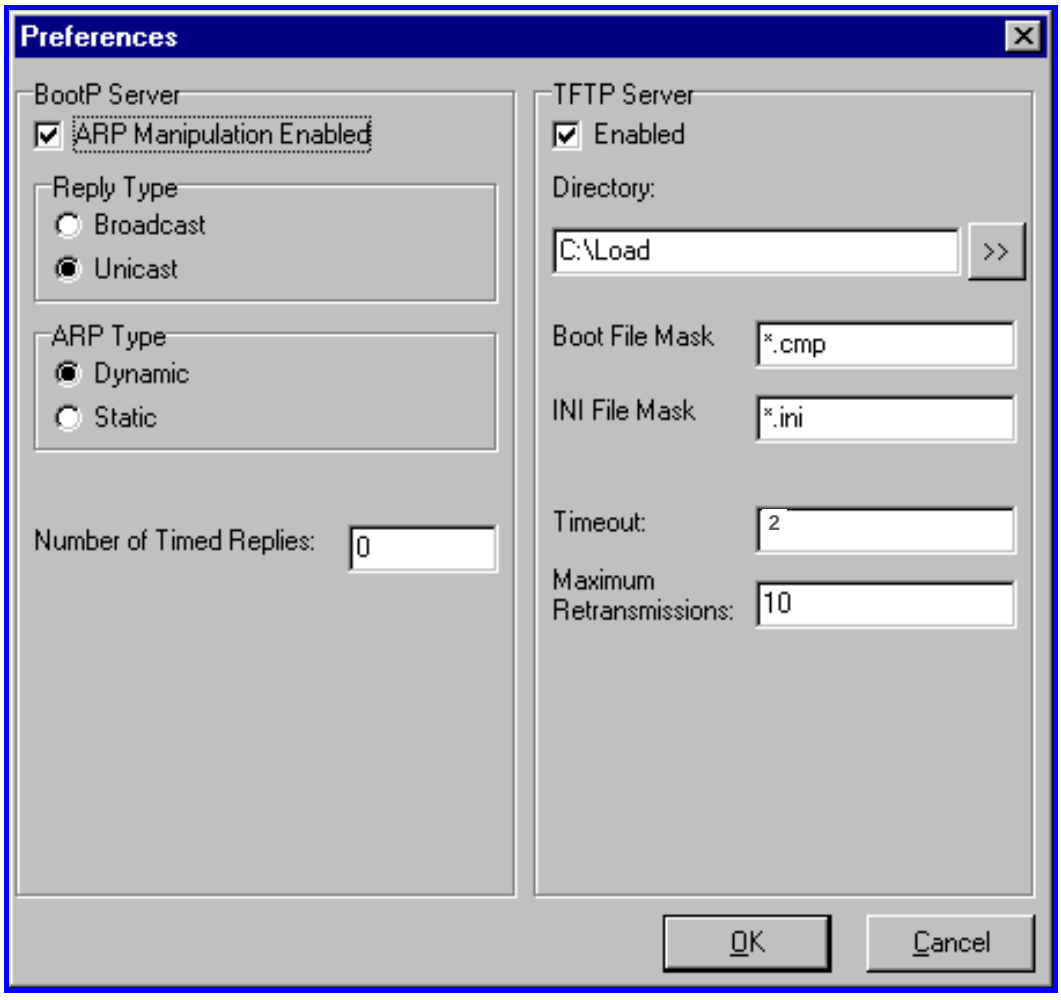
Pop-Up Menu - When the User right-clicks on a line in the log window, the pop-up menu opens. In this menu there two options:

- **Reset**: When this option is selected, the program searches the database for the selected MAC. When the client is found, the program adds the client's MAC to the ARP table, and then sends a reset command to the client. Note that by performing the remote reset this way, the User doesn't have to know the current IP of the client. **To perform this, the User must have administrator privileges**, or else an error message appears.
- **View Client**: This option is the same as double-clicking on a line. When selected, the Clients Window opens. If the Client's MAC is found in the database, it is focused. If not, a new client is added, with the MAC filled out. The User must only fill in the remaining fields.

A.2.2 Preferences Window

In the Preferences window, Figure 10-2, BootP and TFTP configuration parameters are defined.

Figure 10-2: Preferences Window



For the TFTP server, the User can configure a TFTP directory and a value for TFTP Timeout and Maximum Retransmissions. Set these values to 2 and 10 as shown above.

The TFTP server can be disabled by clearing the Enable checkbox.

In the BootP section, the User can select ARP mode: Dynamic or Static, and reply type: Broadcast or Unicast. For a typical application, use Dynamic ARP mode and Unicast, as shown above.

This option requires the **User to have administrator privileges** otherwise an error message appears**.** If you don't have administrator privileges**, uncheck** the ARP Manipulation Enabled checkbox in the Preferences Window, Figure 10-2.

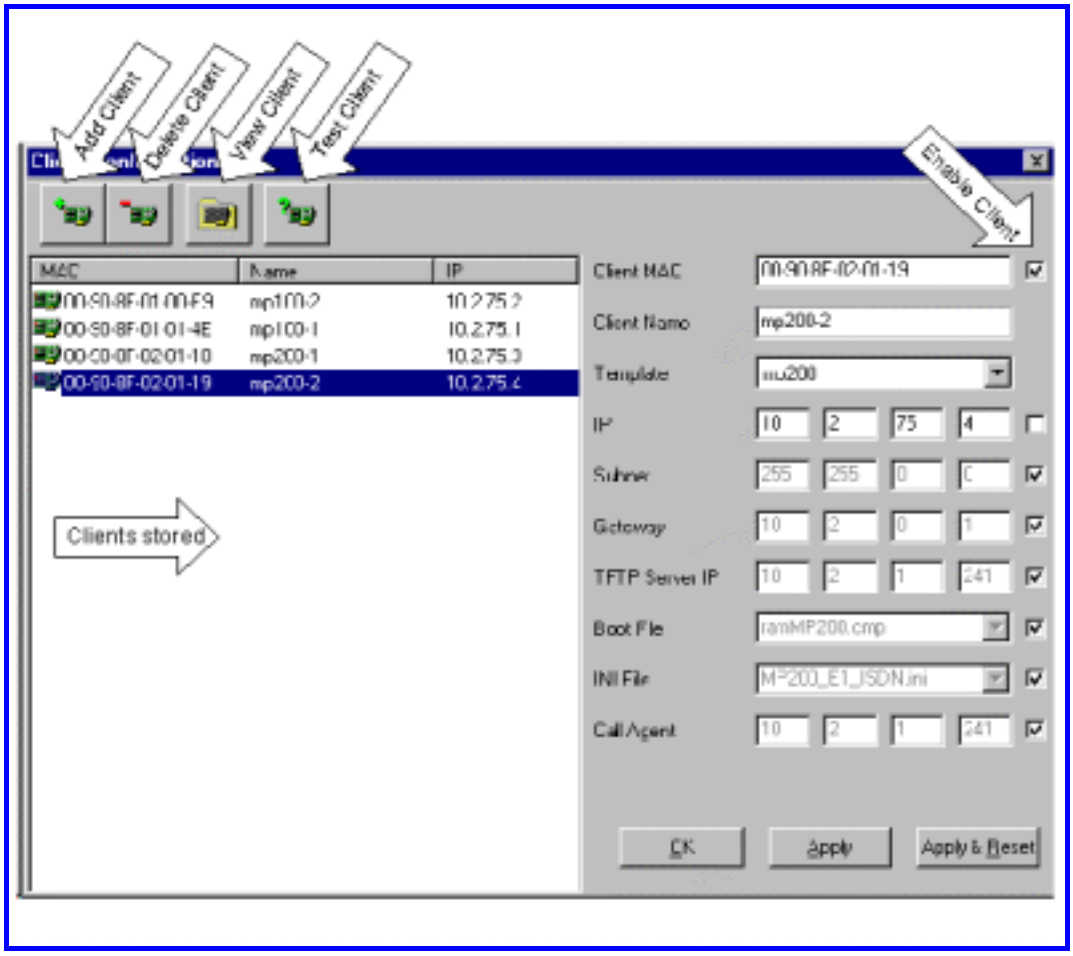
The "Number of Timed Replies" (the number of initiated timed BootP replies) can be used when an MP-1xx is installed behind a Firewall that blocks BootP broadcast requests. In a typical application, this feature can be disabled by entering "0" in this field. When selected, several

BootP replies are sent to the MP-1xx immediately after the remote reset command.

A.2.3 Client Configuration Window

Figure 10-3 is the Client Configuration Window in which clients are added and defined.

Figure 10-3: Client Configuration Window



In the left pane of the window is the client list. By clicking on a client in this list, the following parameters for this client are displayed on the right side of the window:

Client MAC: This is the MAC address of the client. When the User edits the MAC, a new client is added, with the same parameters as the previous client.

The client can be **disabled by un-checking** the check box on the right side of the Client MAC, causing the BootP server not to reply to the BootP request. The client can be **enabled by checking** the check box. Click on the 'Apply' button **each time** the client enable check box is checked or unchecked.

Client Name: Free text for client description.

Template: The template to be used for this client. When a template is selected, its parameters override all of the previous parameters.

IP, Subnet, Gateway: Normal IP parameters.

TFTP Server IP: The IP of the TFTP Server.

Boot File, *ini* File: The files to request from the TFTP server.

Call Agent: The IP of the MGCP Call Agent that will be controlling the Gateway.

Note the seven check boxes to the right of the parameters. These enable the User to assign only

the selected fields from the template. The rest can be unique for each client. When the field is assigned a value from the selected template, the field is grayed out.

After performing changes, press "Apply" to save them. By clicking on "Apply & Reset", the program saves the changes to the database, performs a remote reset to the client by adding the client's MAC to the ARP table, and then sends out a reset command. This option works **only if "ARP Manipulation Enabled"** checkbox in the "Preferences" window is **checked** (in Figure 10-2 on page 115) otherwise an error message appears. It requires the User to have **administrator privileges**. The remote reset is supported for software in this version and up.

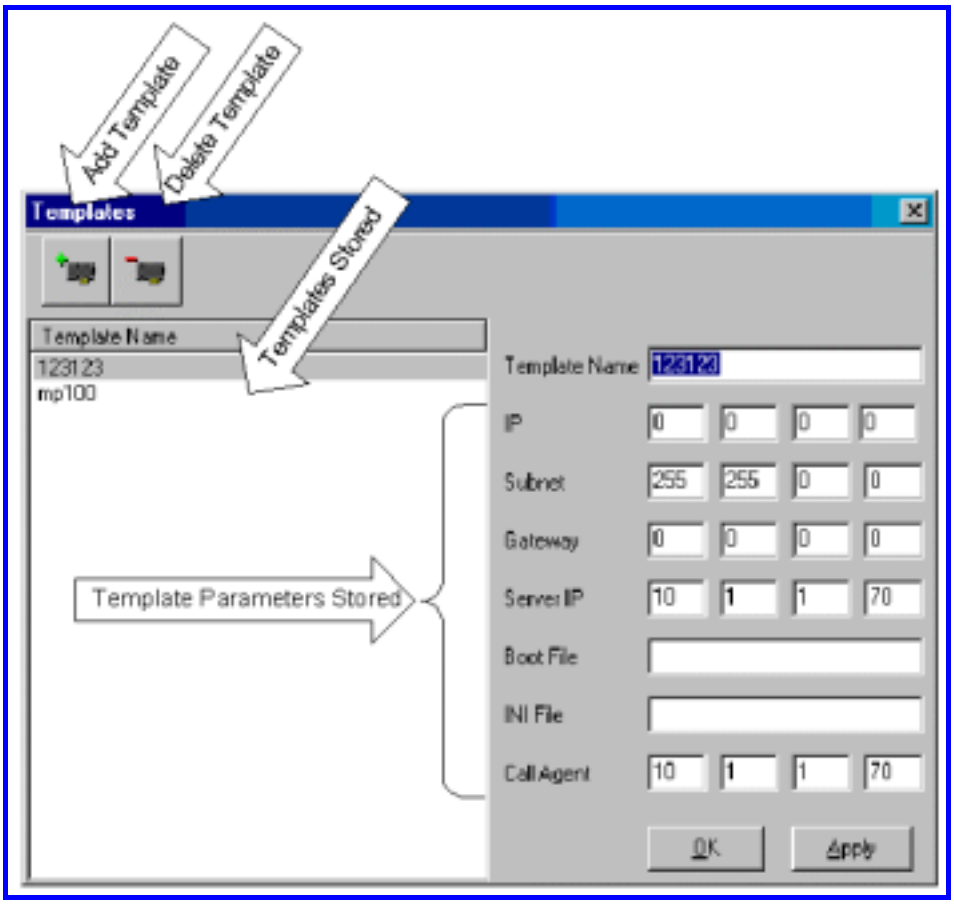
When adding a new client, click on the "Add Client" button; a client with blank parameters is displayed. After filling out the parameters, click on "Apply"; the client is added.

To find out if there is another BootP server on the net that contains a client with the same MAC address, click on "Test Selected Clients"; in the log screen, view the IP addresses of all BootP servers that contain the same MAC address in the status 'Listed At'. In normal operation, BootP client MAC address should be listed only on a single BootP server. If the MAC address is listed in multiple BootP servers, it must be removed from other BootP servers.

A.2.4 Templates Window

Figure 10-4 shows the Templates window, which provides a fast way to configure a number of clients that have the same parameters (except for the IP address). To use the Templates window, create a template, and then apply the template to the client by selecting it.

Figure 10-4: Templates Window



Appendix B - RTP/RTCP Payload Types

RTP Payload Types are defined in RFC 1889 and RFC 1890. AudioCodes has added new payload types to enable advanced use of other coder types. These types are reportedly not used by other applications.

Note: Refer to the IPmedia 2000 Release Notes, Document #: LTRT- 00690 for the supported coders.

B.1 Packet Types Defined in RFC 1890

Table 10-1: Packet Types Defined in RFC 1890

Payload Type	Description	Basic Packet Rate [msec]
0	G.711 µ-Law	20
2	G.726-32	20
4	G.723 (6.3/5.3 kbps)	30
8	G.711 A-Law	20
18	G.729	20
200	RTCP Sender Report	Randomly, approximately every 5 sec (when packets are sent by channel)
201	RTCP Receiver Report	Randomly, approximately every 5 sec (when channel is only receiving)
202	RTCP SDES packet	
203	RTCP BYE packet	
204	RTCP APP packet	

B.2 AudioCodes Defined Payload Types

Table 10-2: AudioCodes Defined Payload Types

Payload Type	Description	Basic Packet Rate [msec]
35	G.726 16 kbps	20
36	G.726 24 kbps	20
38	G.726 40 kbps	20
39	G.727 16 kbps	20
40	G.727 24-16 kbps	20
41	G.727 24 kbps	20
42	G.727 32-16 kbps	20
43	G.727 32-24 kbps	20
44	G.727-32 kbps	20
45	G.727 40-16 kbps	20
46	G.727 40-24 kbps	20
47	G.727 40-32 kbps	20
51	NetCoder 6.4 kbps	20
52	NetCoder 7.2 kbps	20
53	NetCoder 8.0 kbps	20
54	NetCoder 8.8 kbps	20
56	Transparent PCM	20
100	DTMF relay	20
101	Fax Relay	Different packet rates
102	Fax Bypass	20
103	Modem Bypass	20
104	RFC 2198 (Redundancy)	Same as channel's voice coder.

B.3 Default RTP/RTCP/T.38 Port Allocation

The following table shows the Default RTP/RTCP/T.38 Port Allocation.

Table 10-3: Default RTP/RTCP/T.38 Port Allocation

Channel Number	RTP Port	RTCP Port	T.38 Port
1	4000	4001	4002
2	4010	4011	4012
3	4020	4021	4022
4	4030	4031	4032
5	4040	4041	4042
6	4050	4051	4052
7	4060	4061	4062
8	4070	4071	4072
9	4080	4081	4082
10	4090	4091	4092
11	4100	4101	4102
12	4110	4111	4112
13	4120	4121	4122
14	4130	4131	4132
15	4140	4141	4142
16	4150	4151	4152
17	4160	4161	4162
18	4170	4171	4172
19	4180	4181	4182
20	4190	4191	4192
21	4200	4201	4202
22	4210	4211	4212
23	4220	4221	4222
24	4230	4231	4232

Note the changed port allocation from earlier releases, for channel #5 and above.

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Appendix C - DTMF, Fax & Modem Modes

C.1 DTMF Relay Settings

Users can control the way DTMF digits are transported to the remote endpoint, using the DTMFTransport configuration parameter. The following four modes are supported:

DTMFTransportType= 0 (MuteDTMF). In this mode DTMF digits are erased from the audio stream and are **not** relayed to the remote side. Instead silence is sent in the RTP stream.

DTMFTransportType= 1 (Proprietry DTMF Relay). In this mode, DTMF digits are erased from the audio stream and **are** relayed to the remote side using a proprietary RTP syntax.

DTMFTransportType= 2 (Transparent DTMF). In this mode DTMF digits are **left** in the audio stream and the DTMF relay is disabled.

DTMFTransportType= 3 (RFC 2833 DTMF Relay). In this mode, DTMF digits are relayed to the remote side using the RFC 2833 Relay syntax.

C.2 Fax/Modem Settings

Users may choose to use one of the following transport methods for Fax and for each modem type (V.21/V.22/V.23/Bell/V.32/V.34):

- **Fax relay** demodulation / modulation
- **Bypass** using a high bit rate coder to pass the signal
- **Transparent** passing the signal in the current voice coder

When the fax relay mode is enabled, distinction between fax and modem is not immediately possible at the beginning of a session. The channel is therefore in "Answer Tone" mode until a distinction is determined. The packets being sent to the network at this stage are fax relay packets. (The packets can be either T.38-complaint, or FRF.11-based proprietary syntax, selected by setting the channel's configuration parameter UseT38orFRF11.)

C.3 Configuring Fax Relay Mode

When FaxTransportType= 1 (relay mode), then on detection of fax the channel automatically switches from the current voice coder to answer tone mode, and then to fax relay mode. The UseT38orFRF11 configuration parameter defines either T.38-compliant network packets orr proprietary FRF.11-based packets (the last mode should be used mostly for backward-compatibility with previous software versions).

When Fax transmission has ended, the reverse switching from fax relay to voice is performed. This mode switching automatically occurs at both the local and remote endpoints.

Users can limit the fax rate using the FaxRelayMaxRate parameter and can enable/disable ECM fax mode using the FaxRelayECMEnable parameter.

When using T.38 mode, the User can select between two protection strategies – redundancy packets or forward error correction (FEC). This selection is made using the T38FaxRelayProtectionMode configuration parameter. The User can also control a special (proprietary) redundancy mode that was specially designed to improve protection against packet loss using the EnhancedFaxRelayRedundancyDepth parameter. Although this is a proprietary redundancy scheme, it is compatible with other T.38 decoders. When using FRF.11 mode, only redundancy packets are supported. The depth of the redundancy in both protocols (that is, the number of repetitions) is defined by the FaxRelayRedundancyDepth configuration parameter.

Note: T.38 mode currently supports only the T.38 UDP syntax.

C.4 Configuring Fax/Modem ByPass Mode

When VxxTransportType= 2 (FaxModemBypass, Vxx can be one of the following: V32/V22/V21/Bell/V34/Fax), then on detection of Fax/Modem, the channel automatically switches from the current voice coder to a high bit-rate coder, as defined by the User, with the FaxModemBypassCoderType configuration parameter.

If relay is enabled for one of the modes (Fax/Modem), then the Answer Tone mode packets are relayed as fax relay packets.

During the bypass period, the coder uses the packing factor (by which a number of basic coder frames are combined together in the outgoing WAN packet) set by the User in the FaxModemBypassM configuration parameter. The network packets generated and received during the bypass period are regular voice RTP packets (per the selected bypass coder) but with a different RTP Payload type.

When Fax/Modem transmission ends, the reverse switching, from bypass coder to regular voice coder, is carried out.

Note: When Fax relay is enabled, V21TransportType must be set to disable (Transparent) mode.

C.5 Supporting V.34 Faxes

V.34 faxes don't comply with the T.38 relay standard. AudioCodes therefore provides the optional modes described under sections C.5.1 andC.5.2:

Note that the CNG detector is disabled (CNGDetectorMode=0) in all the following examples.

C.5.1 Using Bypass Mechanism for V.34 Fax Transmission

In this proprietary scenario, the Media Gateway uses a high bit-rate coder to transmit V.34 faxes, enabling the full utilization of its speed.

Refer to the following configurations:

FaxTransportMode = 2 (Bypass)
V34ModemTransportType = 2 (Modem bypass)
V32ModemTransportType = 2
V23ModemTransportType = 2
V22ModemTransportType = 2
V21ModemTransportType = 2

In this configuration, both T.30 and V.34 faxes work in Bypass mode.

Or

FaxTransportMode = 1 (Relay)
V34ModemTransportType = 2 (Modem bypass)
V32ModemTransportType = 2
V23ModemTransportType = 2
V22ModemTransportType = 2
V21ModemTransportType = 2

In this configuration, T.30 fax uses T.38 Relay mode while V.34 fax uses Bypass mode.

C.5.2 Using Relay mode for both T.30 and V.34 faxes

In this scenario, V.34 fax machines are compelled to use their backward compatibility with T.30 faxes; as a T.30 machine, the V.34 fax can use T.38 Relay mode.

Refer to the following configuration:

FaxTransportMode = 1 (Relay)
V34ModemTransportType = 0 (Transparent)
V32ModemTransportType = 0
V23ModemTransportType = 0
V22ModemTransportType = 0
V21ModemTransportType = 0

Both T.30 and V.34 faxes use T.38 Relay mode. This configuration forces the V.34 fax machine to operate in the slower T.30 mode.

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Appendix D- DHCP Server Configuration

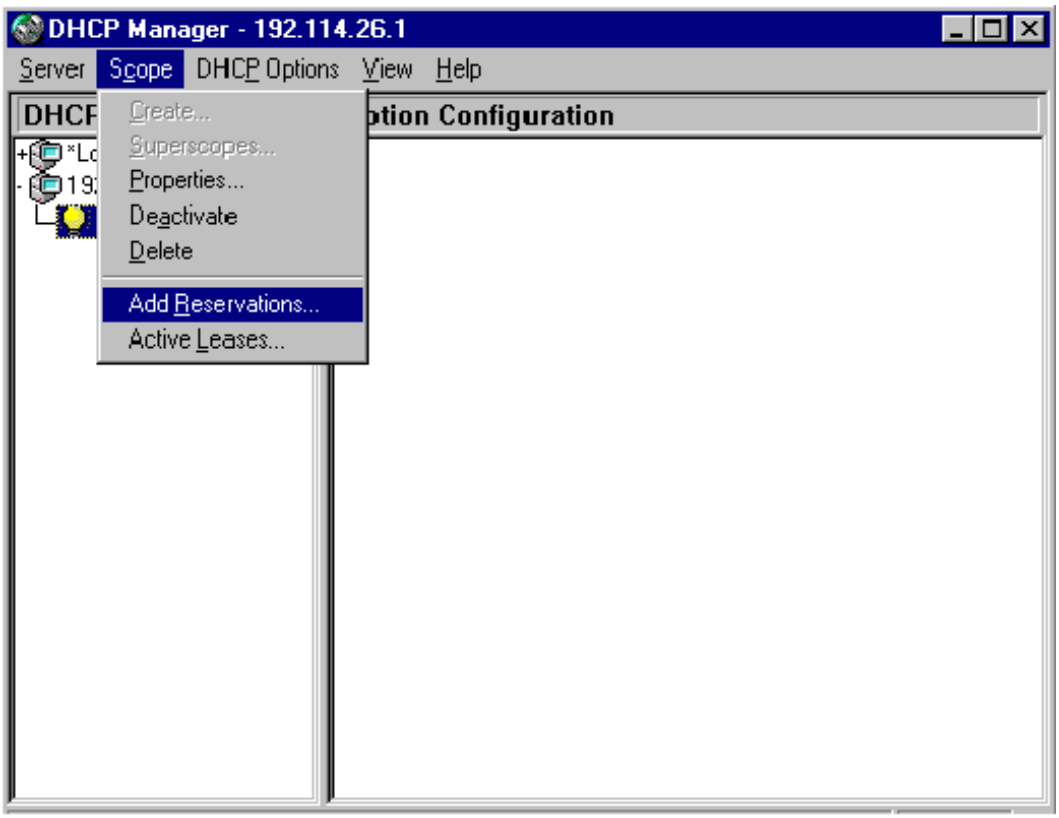
Windows™ NT DHCP Server Configuration in BootP Compatible (Reserve) Mode

Note: The User needs to install Windows™ NT4 services pack 4 (or higher) after enabling the DHCP server service on the NT server. This is required for correct operation with BootP clients.

➢ To configure the Windows™ NT DHCP server take the next 16 steps:

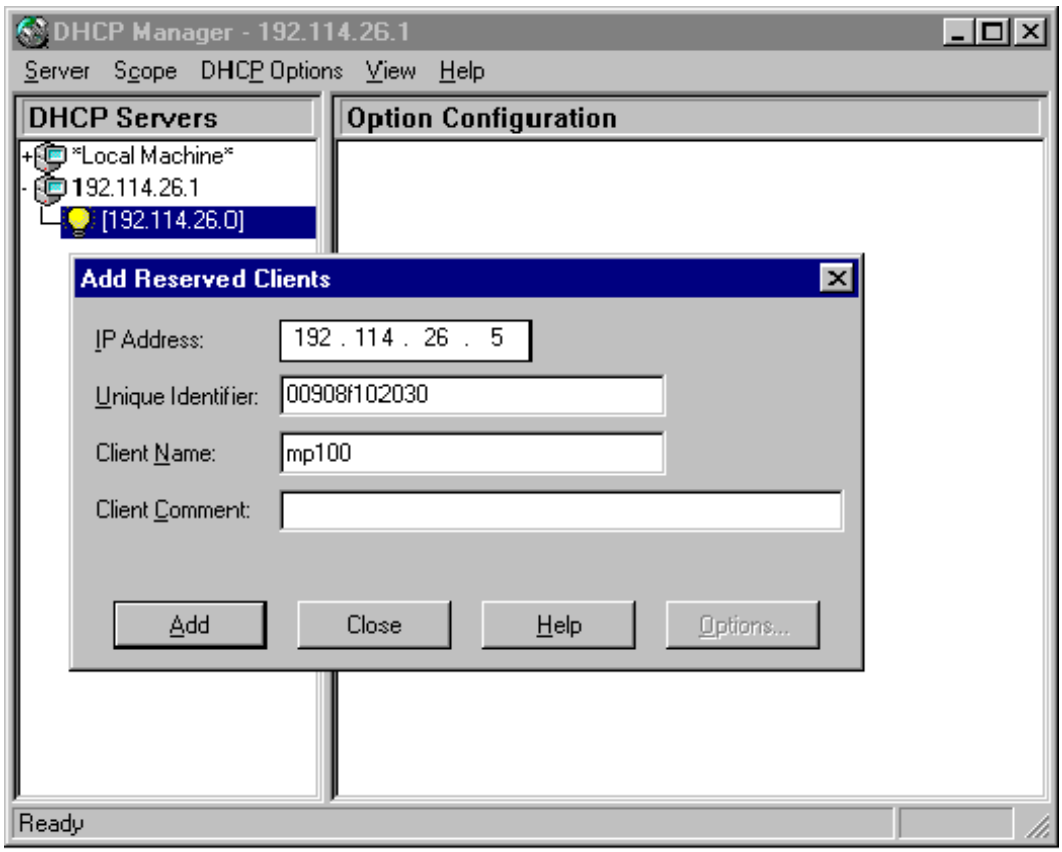
After starting the system, DHCP Manager displays the following screen:

Figure 10-5: Scope Menu



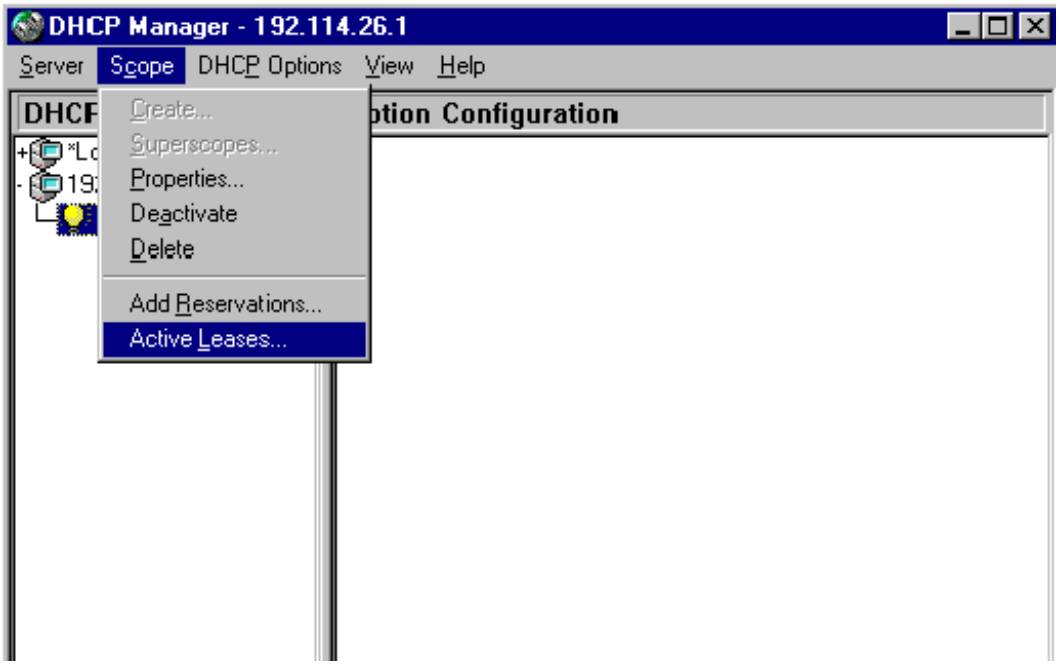
1. Select "Add Reservations" from the "Scope" menu; the 'Add Reserved Clients' screen opens.

Figure 10-6: Add Reserved Clients



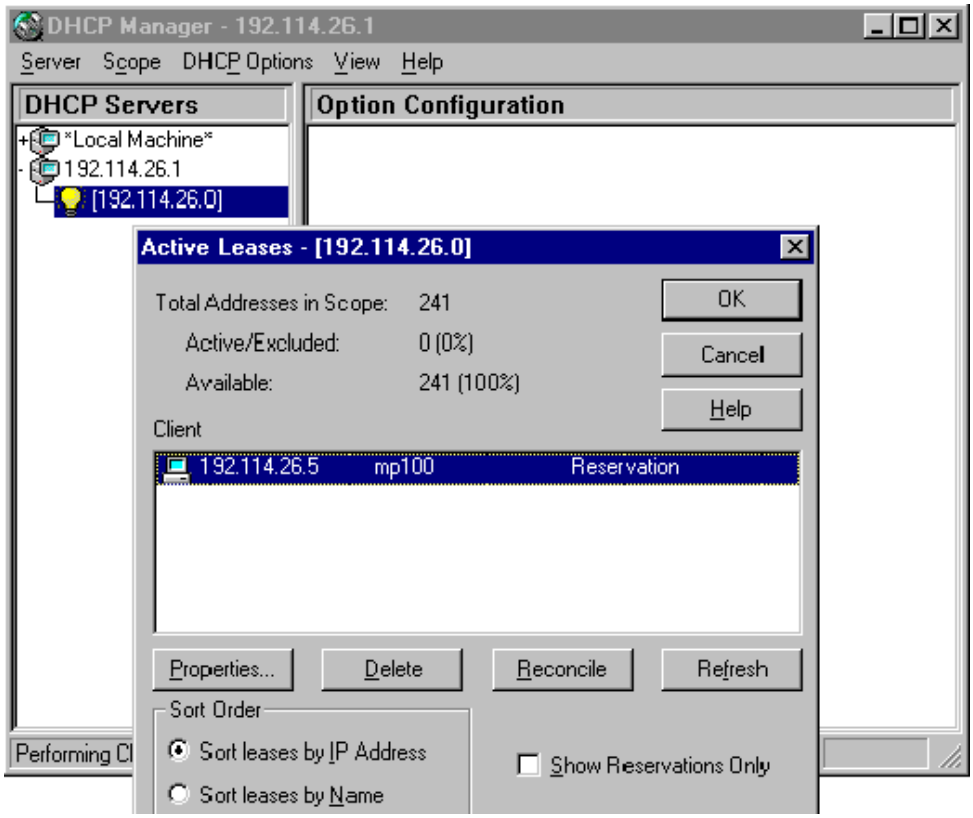
2. Enter the IP address you want to provide to the AudioCodes Media Gateway.
3. The IP address reservation should be inside your DHCP Scope.
4. Enter the hardware MAC address, 12 digits, from your AudioCodes Gateway.
5. Enter Client name; it can be any free text.
6. Select “Add” and then “Close” button.
7. From Scope menu in the main screen, select “Active Leases”.

Figure 10-7: Active Leases Select Screen



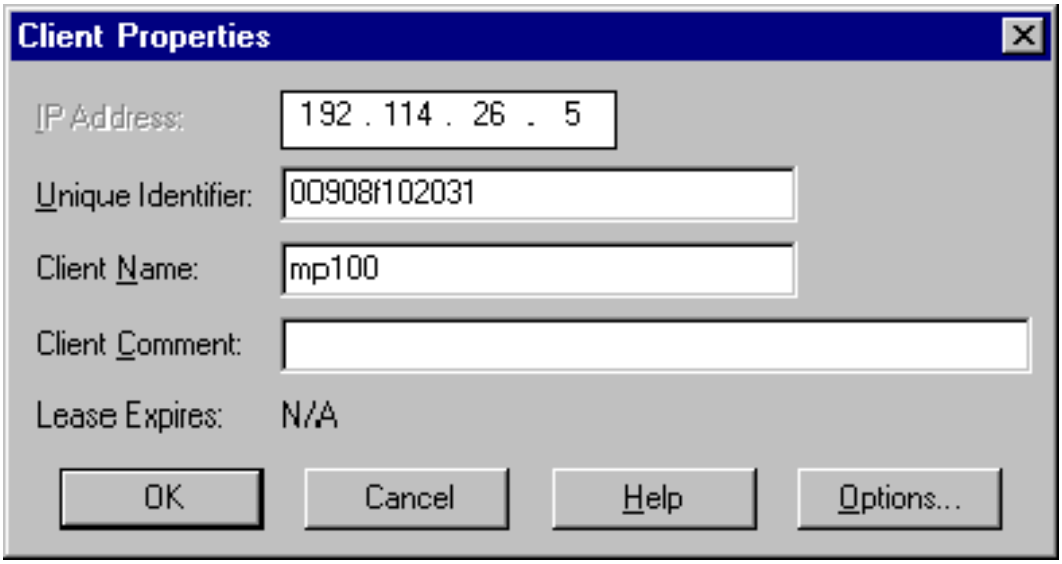
The Active Leases screen displays the following selection box:

Figure 10-8: Active Leases Selection Box



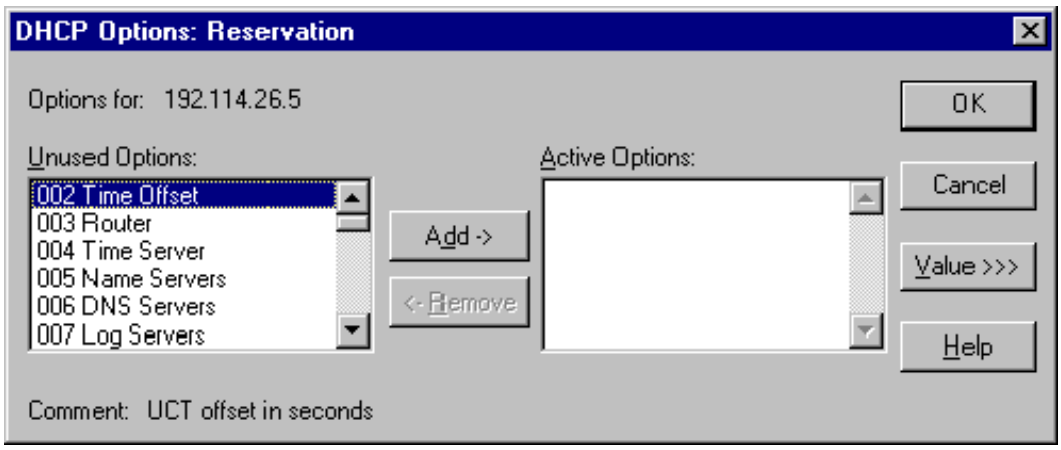
8. Press the Properties button.

Figure 10-9: Client Properties Screen



9. Press the "Options" button to display the 'DHCP Options: Reservation' screen.

Figure 10-10: DHCP Options: Reservation Screen.



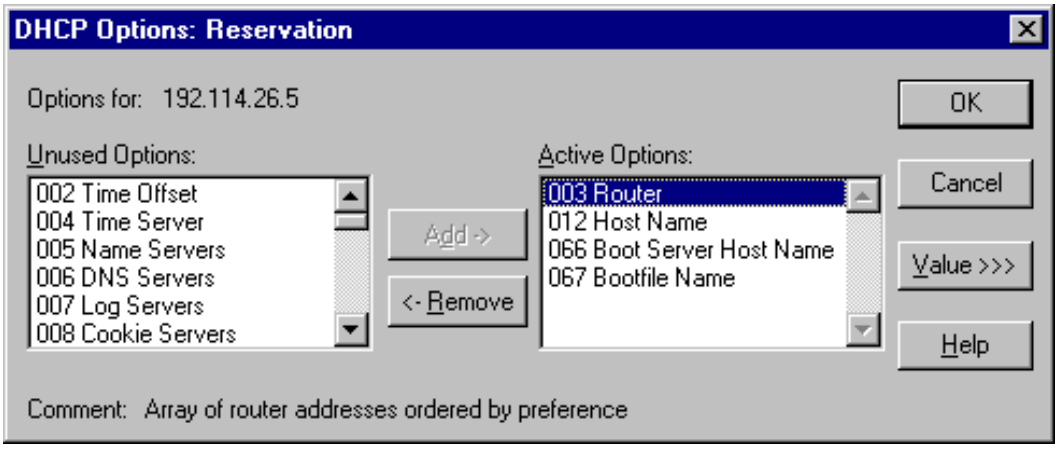
10. Add the following extension fields from the list in the left pane of the 'DHCP Options: Reservation' screen:

003 Router	(Default router/Gateway)
012 Host Name	(BootP client name such as **MP-124** or other Customer-selected name)
066 Boot Server Host Name	(Domain name or IP address of the TFTP server)
067 Bootfile Name	(Such as "ram.cmp")

Note: The file **ram.hex** is a loadable software image file. The file **ram.cmp** is a compressed version of the ram.hex, enabling faster download and reduced file size. The ram.cmp is the only version that can be burned to the flash memory.

The selected extension fields are then displayed in the right pane of the 'DHCP Options: Reservation' screen under 'Active Options', as shown below:

Figure 10-11: Active Options



11. Select the "003 Router" field, as shown above, and click on the "Value" button.
12. Edit the IP Address Array in the 'IP Address Array Editor' screen.

Reader's Notes

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